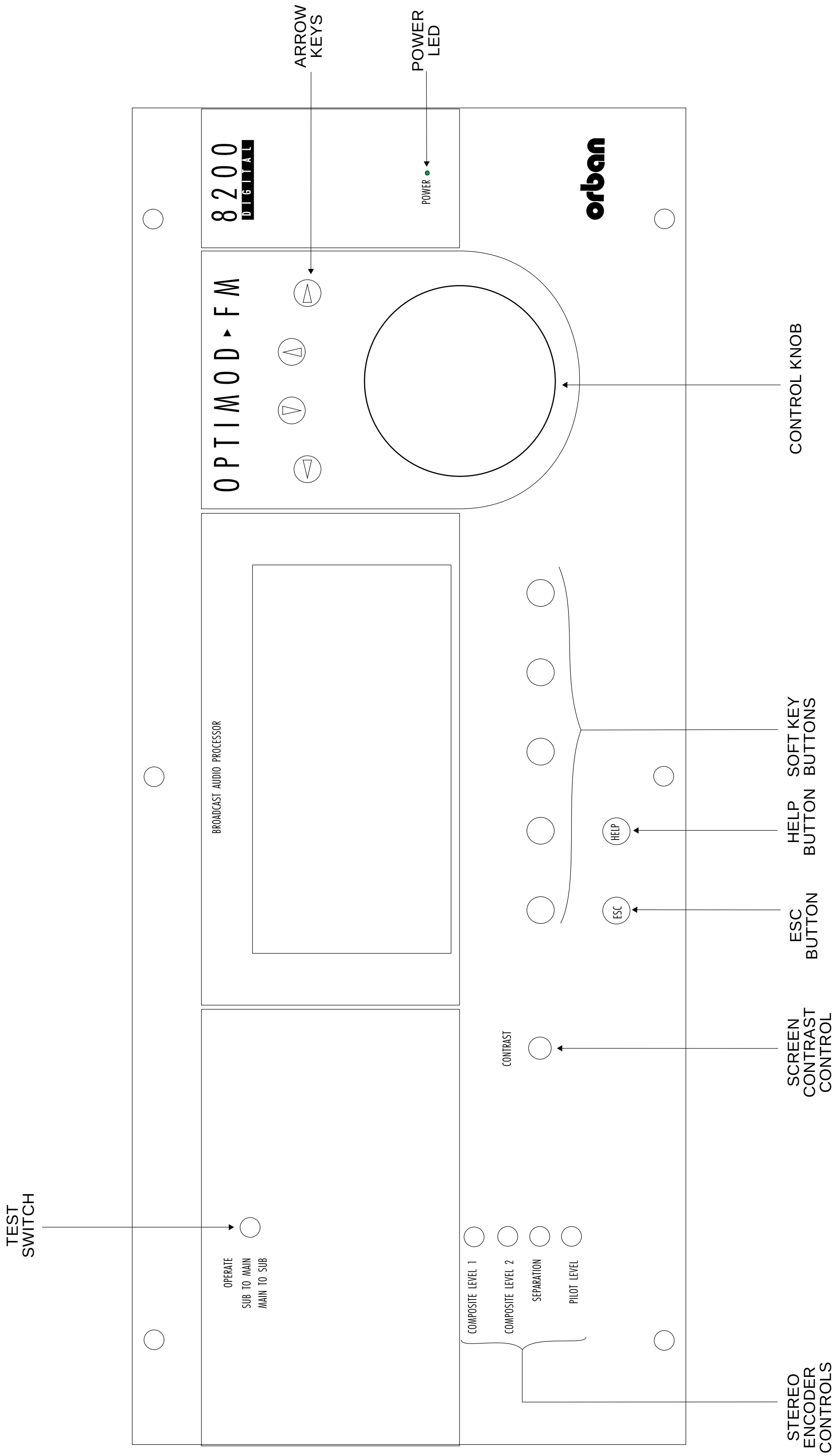
Section 3
Operation

page contents

Caution

The installation and servicing instructions in this manual are for use by qualified personnel only. To avoid electric shock do not perform any servicing other than that contained in the Operating Instructions unless you are qualified to do so. Refer all servicing to qualified service personnel.


ARROW KEYS
POWER LED
8200 DIGITAL
POWER
orban
OPTIMOD-FM
CONTROL KNOB
BROADCAST AUDIO PROCESSOR
SOFT KEY BUTTONS
HELP
HELP BUTTON
ESC
ESC BUTTON
CONTRAST
SCREEN CONTRAST CONTROL
TEST SWITCH
OPERATE
SUB TO MAIN
MAIN TO SUB
COMPOSITE LEVEL 1
COMPOSITE LEVEL 2
SEPARATION
PILOT LEVEL
STEREO ENCODER CONTROLS

8200 Controls and Meters

Arrow Keys (↑, ↓, ←, →) are used for moving around screen to select a function to be changed.

Control Knob is used for changing the setting that is selected by the arrow keys.

Screen Display provides all metering information, labels the five soft-key buttons, provides control setting information.

Screen CONTRAST adjusts the optimum viewing angle of the screen display.

Five Soft Key Buttons provide access to all 8200 functions and controls. The functions of the buttons change with each screen, according to the labels at the bottom of each screen

ESC Button provides an escape from current screen, returns user to the next previous screen, and repeated ESC commands will always return you to the `IDLE G/R` screen.

HELP Button provides HELP information for the current screen and provides detailed help for all of the buttons on that screen (by pushing HELP button, then the screen button).

POWER LED lights when the unit is powered. (It monitors the unregulated +12V DC bus.)

Stereo Encoder Screwdriver-Adjustable Controls

> Orban supplies a special green-handled flat-blade screwdriver (Xcelite R3323) to adjust the stereo encoder controls. Note that the Orban tweaker tool supplied with the analog OPTIMODs cannot be used with the 8200.

Test Switch (OPERATE — SUB TO MAIN — MAIN TO SUB) sets the stereo encoder to operate normally, or to produce pure L–R or L+R signals for system testing.

COMPOSITE LEVEL 1 sets the output level of Composite Output 1.

COMPOSITE LEVEL 2 sets the output level of Composite Output 2.

SEPARATION adjusts the level of the L+R signal, enabling you to optimize the separation through the entire transmission system. (See step 4 on page 4-11.)

PILOT LEVEL adjusts the level of the 19kHz stereo pilot tone. See the note on page 4-14 about accessing and using the pilot meter.

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Introduction to Processing

Some Audio Processing Concepts

Loudness is increased by reducing the peak-to-average ratio of the audio. If peaks are reduced, the average level can be increased within the permitted modulation limits. The effectiveness with which this can be accomplished without introducing objectionable side effects (such as clipping distortion) is the single best measure of audio processing effectiveness.

Compression reduces the difference in level between the soft and loud sounds to make more efficient use of permitted peak level limits, resulting in a subjective increase in the loudness of soft sounds. It *cannot* make loud sounds seem louder. Compression reduces dynamic range relatively slowly in a manner similar to riding the gain: limiting and clipping, on the other hand, reduce the short-term peak-to-average ratio of the audio.

Limiting increases audio density. Increasing density can make loud sounds seem louder, but can also result in an unattractive busier, flatter, or denser sound. It is important to be aware of the many negative subjective side effects of excessive density when setting controls that affect the density of the processed sound.

Clipping sharp peaks does not produce any audible side effects when done moderately. Excessive clipping will be perceived as audible distortion.

Distortion in Processing

In a competently-designed processor, distortion occurs only when the processor is clipping peaks to prevent the audio from exceeding the peak modulation limits of the transmission channel. The less clipping that occurs, the less likely that the listener will hear distortion. However, to reduce clipping, you must decrease the drive level to the clipper, which causes the average level (and thus, the loudness) to decrease proportionally.

The FM pre-emphasis curve introduces further complications. Pre-emphasis boosts the treble at 6dB/octave starting at 2.1kHz (for 75µs countries) or 3.2kHz (for 50µs countries). This reduces the headroom available at high frequencies, and makes it difficult to achieve a bright sound. This is because bright sound requires considerable high frequency power to appear at the output of the receiver's de-emphasis filter, and thus requires a *very large* amount of high frequency power to be transmitted so that a sufficient amount will survive the de-emphasis process.

Without very artful processing, the pre-emphasis will radically increase the level of the peaks and force you to decrease the average level proportionally. Orban's high frequency limiting and distortion-canceling clipping systems greatly ease this trade-off, but cannot entirely eliminate it. Therefore, you can only increase brightness by reducing average modulation (loudness) — unless you accept the increased distortion caused by driving the final clippers harder.

Loudness, Brightness and Distortion.

In processing, there is a *direct trade-off* between loudness, brightness, and distortion. You can improve one only at the expense of one or both of the other two. Thanks to Orban's psychoacoustically-optimized designs, this is less true of Orban processors than of any others. Nevertheless, all intelligent processor designers must acknowledge and work within the laws of physics as they apply to this trade-off.

Perhaps the most difficult part of adjusting a processor is determining the best trade-off for a given situation. We feel that it is usually wiser to give up ultimate loudness to achieve brightness and low distortion. A listener can compensate for loudness by simply adjusting the volume control. But there is *nothing* the listener can do to make an excessively-clipped signal sound clean again, or to undo the effects of excessive high-frequency limiting.

If processing for high quality is done carefully, the sound will also be excellent on small radios. Although such a signal might fall slightly short of ultimate loudness, it will tend to compensate with an openness, depth, and punch (even on small radios) that cannot be obtained when the signal is excessively squashed.

If women form a significant portion of the station's audience, bear in mind that women are more sensitive to distortion and listening fatigue than men. In any format requiring long-term listening to achieve market share, great care should be taken not to alienate women by excessive stridency, harshness, or distortion.

OPTIMOD-FM — from Bach to rock.

OPTIMOD-FM can be adjusted so that the output sounds as close as possible to the input at all times (using the Protection Limiter Structure), or so that it sounds open but more uniform in frequency balance (and often more dramatic) than the input (using any of the Two-Band Structures or slow Multi-Band Structures), or so that it sounds dense, quite squashed, and very loud (using the fast Multi-Band Structure). The dense, loud setup will make the audio seem to jump out of car and table radios, but may be fatiguing and invite tune-outs on higher quality home receivers. The loudness/brightness/distortion trade-off explained above applies to any of these setups.

You will achieve best results if Engineering, Programming, and Management go out of their way to communicate and cooperate with each other. It is important that Engineering understand the sound that Programming desires, and that Management fully understand the trade-offs involved in optimizing one parameter (such as loudness) at the expense of others (such as brightness, distortion, or excessive density).

Never lose sight of the fact that, while the listener can easily control loudness, he or she cannot undo excessive high-frequency limiting or make a distorted signal clean again. If such excessive processing is permitted to audibly degrade the sound of the original program material, the signal is irrevocably contaminated and the original quality can never be recovered.

Customizing the 8200's Sound

The subjective setup controls on the 8200 give you the flexibility to customize your station's sound. But, as with any audio processing system, proper adjustment of these controls consists of balancing the trade-offs between loudness, density, brightness, and audible distortion. The following pages provide the information you need to adjust the 8200 controls to suit your format, taste, and competitive situation.

When you start with one of our factory presets, there are two levels of subjective adjustment available to you to let you customize the factory preset to your requirements:

LESS-MORE

The first level is called LESS-MORE. As you go from less to more, the air sound will become louder, but (as with any processor) processing artifacts will increase. The single LESS-MORE control changes many different subjective setup control settings simultaneously according to a table that we have created in the 8200's permanent ROM (Read-Only Memory). In this table are sets of subjective setup control settings that provide, in our opinion, the most favorable tradeoff between loudness, density, brightness, and audible distortion for a given amount of processing. We believe that most 8200 users will never need to go beyond the LESS-MORE level of control, because the combinations of subjective setup control settings produced by this control have been optimized by Orban's audio processing experts on the basis of years of experience designing audio processing, and upon hundred of hours of listening tests.

Please note that the highest LESS-MORE setting is purposely designed to cause unpleasant distortion and processing artifacts! This helps assure you that the setting of the LESS-MORE control that you choose is optimum, because turning the control up to this point will cause the sound quality to become obviously unacceptable.

FULL-CONTROL

If you want to create a *signature sound* for your station that is out of the ordinary, or if your taste differs from the people who programmed the LESS-MORE tables, FULL-CONTROL is available to you. At this level, you can customize or modify any subjective setup control setting to create a sound exactly to your taste. You can then save the settings in a user preset and recall it whenever you wish.

You need not (in fact, *cannot*) create a sound entirely from scratch. All user presets are created by modifying factory presets or by further modifying factory presets that have been previously modified with a LESS-MORE adjustment. Because the LESS-MORE control also adjusts certain invisible system parameters that cannot be accessed by the user even at the FULL-CONTROL level, it is wise to set the LESS-MORE control to achieve a sound as close as possible to your desired sound before you make further modifications at the FULL-CONTROL level.

If you want to create your own user presets, the following detailed discussion of the processing structures is important to understand.

If you only use factory presets, or if you only modify them with LESS-MORE, then you may still find the material interesting, but it is not necessary to understand it to get excellent sound from the 8200.

Fundamental Requirements: High-Quality Source Material and Accurate Monitoring

A major potential cause of distortion is excess clipping. Another cause is poor-quality source material, including the effects of the station's playback machines, electronics, and studio-to-transmitter link. If the source material is even slightly distorted, that distortion can be greatly exaggerated by OPTIMOD-FM — particularly if a large amount of gain reduction is used. Very clean audio can be processed harder without producing objectionable distortion. See *Audio Quality in the FM Plant* (a separate Orban publication included with each unit) for a discussion of how to improve source quality.

A high-quality monitor system is essential. To modify your air sound effectively, you must be able to *hear* the results of your adjustments. In too many stations, the best monitor is significantly inferior to the receivers found in many listeners' homes! See *Audio Quality in the FM Plant* for a detailed discussion of how to efficiently create an accurate monitoring environment (and otherwise bring the audio plant up to state-of-the-art quality).

About the Processing Structures

In the 8200, a *processing structure* is a program that operates as a complete audio processing system. Only one processing structure can be active at a time. Just as there are many possible ways of configuring a processing system using analog components (like equalizers, compressors, limiters, and clippers), there are many possible processing structures achievable by the 8200. Unlike an analog system, where creating a complete processing system involves physically wiring its various components together, the 8200 realizes all of its processing structures as a series of high-speed mathematical computations made by Digital Signal Processing (DSP) integrated circuit chips. So the 8200 can be changed from one structure to another almost instantly by loading new software from high-speed semiconductor memory within the 8200.

Refer to the second page of this manual for the processing structures included in your version.

Factory Programming Presets

Factory Programming Presets are our "factory recommended settings" for various program formats or types. Each Factory *Programming* Preset is simply a Factory *Processing* Preset set to the LESS-MORE setting that is likely to match that program format in an average market.

The Factory Programming Presets are designed as starting points to help you get on the air quickly without having to understand anything about adjusting the 8200's sound. Each of these presets can be edited with the LESS-MORE control to optimize the tradeoff between loudness, brightness, and distortion according to the needs of the format. Because it is so easy to fine-tune the sound at the LESS-MORE level, we think that most users will quickly want to customize their chosen preset to complement their market and competitive position after they had time to familiarize themselves with the 8200's programming facilities.

Start with one of these presets. Spend some time listening critically to your on-air sound. Listen to a wide range of program material typical of your format, and listen on several types of radios (not just on your studio monitors). Then, if you wish, customize your sound using the information in the Protection Limiter, Two-Band and Multi-Band sections that follow.

Factory Programming Presets if Multi-Band Processing not installed (2S Version)

##	FORMAT	FACTORY PROCESSING PRESET	LESS-MORE
FA	CLASSICAL	BB 2B-PUR CLASSICAL	5.0
FB	HARD CHR	CA 2B-NOR PROCESSED	9.0
FC	MEDIUM CHR	CA 2B-NOR PROCESSED	7.5
FD	SOFT CHR	CA 2B-NOR PROCESSED	6.0
FE	MODERN COUNTRY	CA 2B-NOR PROCESSED	7.0
FF	AOR (Album-Oriented Rock)	CA 2B-NOR PROCESSED	7.0
FG	ADULT CONTEMP	CA 2B-NOR PROCESSED	7.0
FH	URBAN	CA 2B-NOR PROCESSED	8.0
FI	SOFT URBAN	CA 2B-NOR PROCESSED	6.0
FJ	NEW AGE	CA 2B-NOR PROCESSED	4.0
FK	OLDIES	CA 2B-NOR PROCESSED	7.0
FL	JAZZ	CA 2B-NOR PROCESSED	4.0
FM	BEAUTIFUL	CA 2B-NOR PROCESSED	4.0
FN	MOR (Middle-of-the-Road)	CA 2B-NOR PROCESSED	5.0
FO	TALK	CA 2B-NOR PROCESSED	7.0

Factory Programming Presets if Multi-Band Processing installed (3S Version)

##	FORMAT	FACTORY PROCESSING PRESET	LESS-MORE
FA	CLASSICAL	BB 2B-PUR CLASSICAL	5.0
FB	HARD CHR	DD 5B-FAST	8.0
FC	MEDIUM CHR	DC 5B-MEDIUM FAST	7.5
FD	SOFT CHR	DB 5B-MEDIUM SLOW	7.0
FE	MODERN COUNTRY	DB 5B-MEDIUM SLOW	7.0
FF	AOR (Album-Oriented Rock)	DA 5B-SLOW	7.0
FG	ADULT CONTEMP	DB 5B-MEDIUM SLOW	5.0
FH	URBAN	DD 5B-FAST	7.0
FI	SOFT URBAN	DA 5B-SLOW	6.0
FJ	NEW AGE	DA 5B-SLOW	4.0
FK	OLDIES	DB 5B-MEDIUM SLOW	6.0
FL	JAZZ	DA 5B-SLOW	5.0
FM	BEAUTIFUL	DA 5B-SLOW	4.0
FN	MOR (Middle-of-the-Road)	DA 5B-SLOW	5.0
FO	TALK	DC 5B-MEDIUM FAST	7.0

FA CLASSICAL produces a very clean, open sound that is ideal for stations that primarily play classical music, whose success depends on attracting and holding audiences for very long periods of time. CLASSICAL produces an unprocessed sound with a nice sense of dynamic range. This preset is based on the Purist Two-Band structure, because classical broadcasters usually want to preserve the essential fidelity and "sound" of the original recording with minimum extraneous coloration. [A station that provides classical music primarily for background listening might want to experiment with the NEW AGE or BEAUTIFUL presets (which are identical). These presets make the program material more consistent than does the CLASSICAL preset. Accordingly, they can ensure that all parts of the music are heard even when it is played at low levels in the background.]

FB HARD CHR (Hard Contemporary Hit Radio): provides processing for stations that primarily play modern Top 40 music. The fast release time of HARD CHR produces a very loud, consistent, dramatic, and rather synthetic sound that is ideal for a highly competitive Contemporary Hit Radio (CHR) format whose ratings depend on attracting a large number of listeners (high "cume") but which does not assume that a listener will listen to the station for hours at a time. This is the major market competitive sound, emphasizing loudness as well as clean audio. The Two-Band HARD CHR preset, with essentially independent operation of the two bands, makes the sound from cut to cut and announcer to announcer more consistent. The Multi-Band HARD CHR preset is remarkably consistent as the texture of music is noticeably altered to a standard. The bass has an ever-present punch, there is always a sense of presence, and highs are in perfect balance to the mids, no matter what was on the original recording.

FC MEDIUM CHR (Medium Contemporary Hit Radio): provides processing for stations that primarily play modern Top 40 music, but where the market is not engaged in loudness wars. MEDIUM CHR provides a high-energy sound, but with less density (and therefore less danger of fatiguing listeners) than HARD CHR.

FD SOFT CHR (Soft Contemporary Hit Radio): is appropriate for stations which play softer Top 40 music, that need a glossy "show-business" sound, yet whose ratings depend on maintaining a longer time spent listening than do other CHR formats. This is the sound texture for the station that values a clean, easy-to-listen-to sound. With Multi-Band SOFT CHR, a tasteful amount of punch, presence, and brightness is added when appropriate.

FE MODERN COUNTRY is appropriate for stations that play primarily modern country music. The ratings of modern country stations usually depend on maintaining a longer time spent listening than do conventional CHR formats. This is the sound texture for the station that values clean, easy-to-listen-to unprocessed audio that sounds just right on music and voice when listened to on small table radios, car radios, portables, or home hi-fi systems. With Multi-Band MODERN COUNTRY, a tasteful amount of punch, presence, and brightness is added when appropriate.

FF AOR (Album-Oriented Rock): produces a very punchy, clean, open sound that is ideal for stations that primarily play Album-Oriented Rock, and whose success depends on attracting and holding audiences for very long periods of time. AOR stations typically play a mix of current, non-dance-oriented rock music and familiar "heritage" cuts from the last twenty years or so. Their listeners tend to be older than the CHR and Urban audience. FF AOR is also suitable for modern rock and alternative formats. FF AOR produces an unprocessed sound with a nice sense of dynamic range. Multi-Band AOR provides gentle automatic equalization to keep the frequency balance consistent from record to record (especially those recorded in different eras).

FG CONTEMPORARY (Adult Contemporary): is appropriate for adult-oriented stations that play soft rock. This is the sound texture for the station that values clean, easy-to-listen-to unprocessed audio that sounds just right on music and voice when listened to on small table radios, car radios, portables, or home hi-fi systems. With Multi-Band CONTEMPORARY, a tasteful amount of punch, presence, and brightness is added when appropriate.

FH URBAN (Urban Contemporary) provides processing for stations that primarily play upbeat urban music from Funk to Rap, and whose audience listens primarily on smaller radios. The fast release time of URBAN produces a very loud, consistent sound that is likely to attract a large number of listeners but which may be fatiguing over a long period of time. Because the high amount of processing tends to squash bass, bass does not have the same amount of punch that it does with the SOFT URBAN preset, and stations catering to "boom boxes" and aftermarket auto radios with exaggerated bass will probably prefer the punchier (although quieter) sound of SOFT URBAN.

FI SOFT URBAN produces a very punchy, clean, open sound. While noticeably quieter than URBAN, it makes up for the loudness difference by producing the punch and slam that modern urban music requires for full impact.

FJ NEW AGE produces an extremely clean, open sound with almost no audible processing. The settings for Multi-Band NEW AGE provide gentle automatic equalization to keep the frequency balance consistent from cut to cut. Even when the radio is played softly, these settings ensure that your sound doesn't lose its highs and lows.

FK OLDIES is appropriate for stations that play music from other eras, from 50's rock 'n' roll to the Beatles. This is the sound texture for the station that values clean, easy-to-listen-to audio. With Multi-Band OLDIES, a tasteful amount of punch, presence, and brightness is added when appropriate. Also, the Multi-Band's compression automatically re-equalizes older material for consistency without losing its essential texture and flavor.

FL JAZZ produces an extremely clean, open sound with almost no audible processing. The settings for Multi-Band JAZZ provide gentle automatic equalization to keep the frequency balance consistent from record to record (especially those recorded in different eras). ["Purist" jazz stations may want to use Two-Band processing instead, particularly if they don't want the processing to re-equalize the music but instead faithfully preserve the "sound" of the original recording. FA CLASSICAL gives this sound.]

FM BEAUTIFUL produces a very open, unprocessed sound. It is ideal for background music formats that play mostly soft instrumental music. This is a sound that is easily listenable for many hours without fatiguing listeners. For 8200's with Multi-Band BEAUTIFUL, the compressor re-equalizes program material as necessary and ensures that all parts of the music are audible, even when the radio is played very quietly.

FN MOR (Middle-of-the-Road) produces an extremely clean, open sound with almost no audible processing. It is ideal for stations playing older pop standards, big-band, and other non-rock music. Its easy-to-listen-to quality holds the older adult audience that usually listens to this format. The settings for Multi-Band MOR provide gentle automatic equalization to keep the frequency balance consistent from record to record (especially those recorded in different eras).

FO TALK provides processing for Talk format stations that primarily feature news, call-in shows, interviews, and other voice material. FO TALK keeps the levels of announcers and guests consistent, and keeps a proper balance between voice and commercials. Voice is the most difficult audio to process, but these settings result in a favorable trade-off between consistency, presence, and distortion.

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The Protection Limiter Structure

The Protection Limiter Structure is designed for stations wanting the highest possible fidelity to the source, such as a station broadcasting concert music at night when its audience is likely to listen in a concentrated and critical way. Unlike the other structures, the Protection Limiter Structure is *not* designed to reduce the dynamic range, to increase program density, to increase loudness, or to increase the consistency of sound from different sources. Its only function is to protect the transmitter from over-deviation while preserving the spectral and dynamic quality of the source material.

The Protection Limiter Structure is based on Orban's analog Model 4000 Transmission Limiter. Like the 4000, it is designed to be operated below the threshold of limiting most of the time. There are virtually no user controls — the parameters of the structure have been chosen to make it audibly *undetectable*.

Unlike the other structures, the maximum gain reduction in the Protection Limiter Structure is 15dB. This is entirely sufficient for the purpose, and optimizes the signal-to-noise ratio available through the system. Further, the group delay through the system is essentially constant throughout the audible frequency range.

There are two factory presets for the Protection Limiter Structure. AA sets the limiting threshold so that limiting almost never occurs, while AB sets the limiting threshold so that program material at the maximum normal input level (as determined by a PPM or VU meter monitoring the input program line) produces an average limiting of 5dB.

Setting Up the Protection Limiter

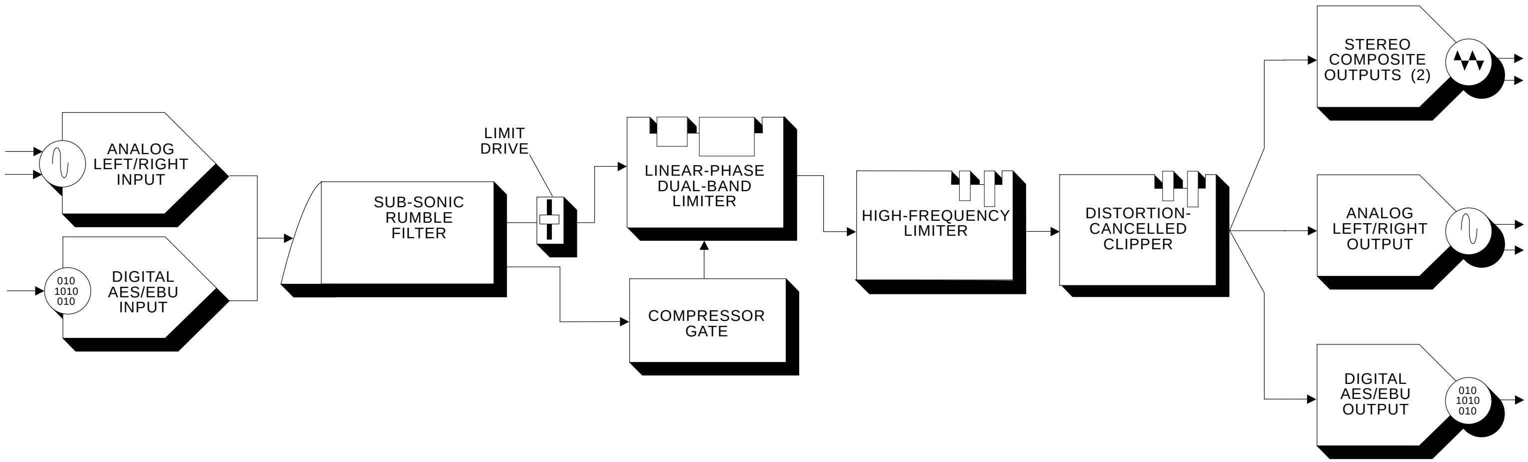
To set up the Protection Limiter, recall factory preset AA (if you want limiting to occur only when the program level exceeds the maximum normal input level as determined by a PPM or VU meter monitoring the input program line), or preset AB (if you want about 5dB of limiting to occur at the maximum normal input level).

The LESS-MORE control affects only the input drive, and you can use it to set a nominal limiting level different than 0dB or 5dB.

There are two FULL-CONTROL parameters for the Protection Limiter.

LIMIT DRIVE LEVEL duplicates the effect of the LESS-MORE control.

30 Hz HPF determines if the 30Hz high-pass filter prior to the limiter is in or out of the signal path.



Protection Limiter Structure
(Simplified Block Diagram)

The 30Hz high-pass filter has an 18dB/octave slope and is down 0.5dB at 30Hz. It eliminates modulation-wasting subsonic energy from acoustic and turntable rumble, and removes most of the energy from *pops* caused by breath blasts into microphones. It prevents any such subsonic energy from modulating the audio processor's AGC and compressor control signals (which could cause unpleasant distortion), and prevents the automatic frequency control loops in FM exciters from introducing modulation distortion into the audio or even becoming entirely unlocked.

The cutoff frequency of this filter is so low that the only common musical instruments producing lower fundamental frequencies are the pipe organ and synthesizer. The bass energy in most pop music occurs above 40Hz. The ringing introduced by the filter is insignificant. The ear is very insensitive to ringing in this frequency range. To put the issue in perspective, the ringing is comparable to that introduced by a well-designed vented box loudspeaker with a 30Hz cutoff. For all of these reasons, we recommend operating the system with the 30Hz HIGHPASS control set ON. We can only recommend setting it OFF if the programming is exclusively from compact disc or tape mastered from non-vinyl sources, and if high-pass filters are applied to all live microphones to prevent pops or rumble from disturbing later components in the broadcast chain.

Protection Limiting: Orban's Approach

Traditionally, protection limiters have used peak-sensing automatic gain control (AGC) processors to control peak levels. Superficially, this approach seems reasonable. The purpose of a protection limiter is to control the peak levels in a transmission channel.

This traditional approach ignores one crucial requirement for protection 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 subjectively-perceived loudness. Audio material with a high peak-to-average ratio emerges from such a limiter much quieter than audio material with a low peak-to-average ratio. The ear perceives this as an unnatural, unpleasant *pumping* quality. Thus the traditional 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*.

The dual-band limiter controls the level driving the following high-frequency limiter and distortion-canceling clipper stages. Prior to the HF limiter, a phase-coherent crossover divides the signal into frequency bands above and below 150Hz. The above-150Hz material is connected to the Master Band, which determines the overall limiting. This prevents limiter-induced spectral gain intermodulation — audible modulation of the loudness of midrange and high frequency program material by bass-generated limiting.

The below-150Hz material is connected to the Bass Band. The gain-control voltage produced by the Master Band is cross-coupled into the Bass Band, so that the gain of the Bass Band ordinarily tracks the gain of the Master Band exactly, preserving frequency balances. When the Bass Band encounters exceptionally heavy bass, it momentarily provides extra limiting to preclude excessive level at the dual-band limiter's output.

The dual-band limiter has an attack time of approximately 2 milliseconds. This moderate attack time prevents it from producing limiting on every transient spike. Such limiting could otherwise create audible *holes* in the program.

The dual-band limiter is gated: when its input level drops below the factory-set *threshold of gating*, the release rate is radically slowed to avoid audible *noise breathing*.

> This compressor gate is not the same as a conventional noise gate because it is not intended to reduce noise or other low-level undesired sounds to a lower level than that occurring in the original program. Its only purpose is to prevent the unnatural exaggeration of such material.

The dual-band limiter can produce a maximum of 15dB of limiting. This is more than adequate for protection limiting. *It is important not to overdrive the limiter past 15dB gain reduction*; the sound will rapidly become highly distorted.

Because the gain control section of the Protection Limiter Structure is not peak-sensing, its output contains peak overshoots that must be eliminated by further processing. The Protection Limiter Structure provides three cascaded processors 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 processing.

The Smart Clipper distortion-canceling clipper is the prime means for peak control. Because it *removes only instantaneous peaks in the waveform that exceed the desired limiting level,* it does not affect the average level and does not cause unnatural loudness variation. Traditionally, clippers cause objectionable distortion. Such distortion is prevented in the Smart Clipper by proprietary, patented processing 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 processor contains low-pass filters that strictly constrain the output spectrum to 15kHz, but which overshoot. These overshoots are eliminated by the Frequency-Contoured Sidechain overshoot corrector. This processor 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 do.

Modulation of Sine waves

There is an important and sometimes confusing consequence of this system design: **the system will not permit sine waves to reach 100% peak modulation**. It will restrain sinewave modulation to a lower level — typically 7dB below 100% (45% modulation). Therefore, in its normal OPERATE mode the Protection Limiter Structure will not pass an externally-generated line-up tone at 100% modulation; it will produce limiting that constrains the tone to approximately 45% modulation.

This is a direct consequence of the level detection's being power-sensing. For a given peak level, sine waves have very high average power by comparison to program material. To preserve natural sound, the processing must reduce their peak level below the peak level of program material to preserve consistent average power at the limiter's output. This is a characteristic of any limiter that achieves natural-sounding dynamic performance and that does not modulate program loudness according to the peak-to-average ratio of the input signal.

Almost all program material will produce frequent peaks at 100% modulation at the 8200's output. Program material that does not produce such peaks has an unusually low peak-to-average ratio and will sound naturally balanced when applied to the transmission system below 100% peak modulation.

Delay-Line Techniques vs. Distortion-canceling Clipping

The Protection Limiter Structure was designed to achieve 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 inconsistent with these goals because a delay-line limiter is simply a highly refined peak-sensing AGC processor. 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. While this effect is somewhat reduced by an automatic release algorithm, 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 input. 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.

More on High-Frequency Limiting

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.

In contrast, the HF limiter in the Protection Limiter Structure operates in several frequency bands and adapts to the spectrum of the program material 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.

Please note that the HF limiter in the Protection Limiter Structure will not exhibit a swept-sinewave frequency response inverse to the pre-emphasis curve set by the Protection Limiter Structure's software setting. This is because the HF limiter, as explained above, does as little work as possible: most of the work in controlling HF overload is done by the distortion-canceling clipper. Be assured that the overall Protection Limiter Structure system is nevertheless operating to very tight tolerances on the pre-emphasis curve.

The Two-Band Structures

There are two Two-Band Structures available: Both consist of a slow single-band gated AGC (Automatic Gain Control) for gain riding, followed by a gated two-band compressor and a high-frequency limiter.

The Two-Band Normal Structure (2B NOR PROCESSED) is an improved version of Orban's classic 8100A OPTIMOD-FM, but with increased high frequency clarity. It is operated after a phase rotator (time-dispersion filter) to improve its loudness capability by making positive and negative peaks more symmetrical, particularly with voice.

The Two-Band Purist Structure (2B-PUR PROCESSED and 2B PUR CLASSICAL) is designed for purists who want constant time delay at all frequencies (linear phase) throughout the 30Hz to 15kHz frequency range. It is capable of about 3dB lower loudness than the Two-Band Normal Structure for a given amount of perceived clipping distortion.

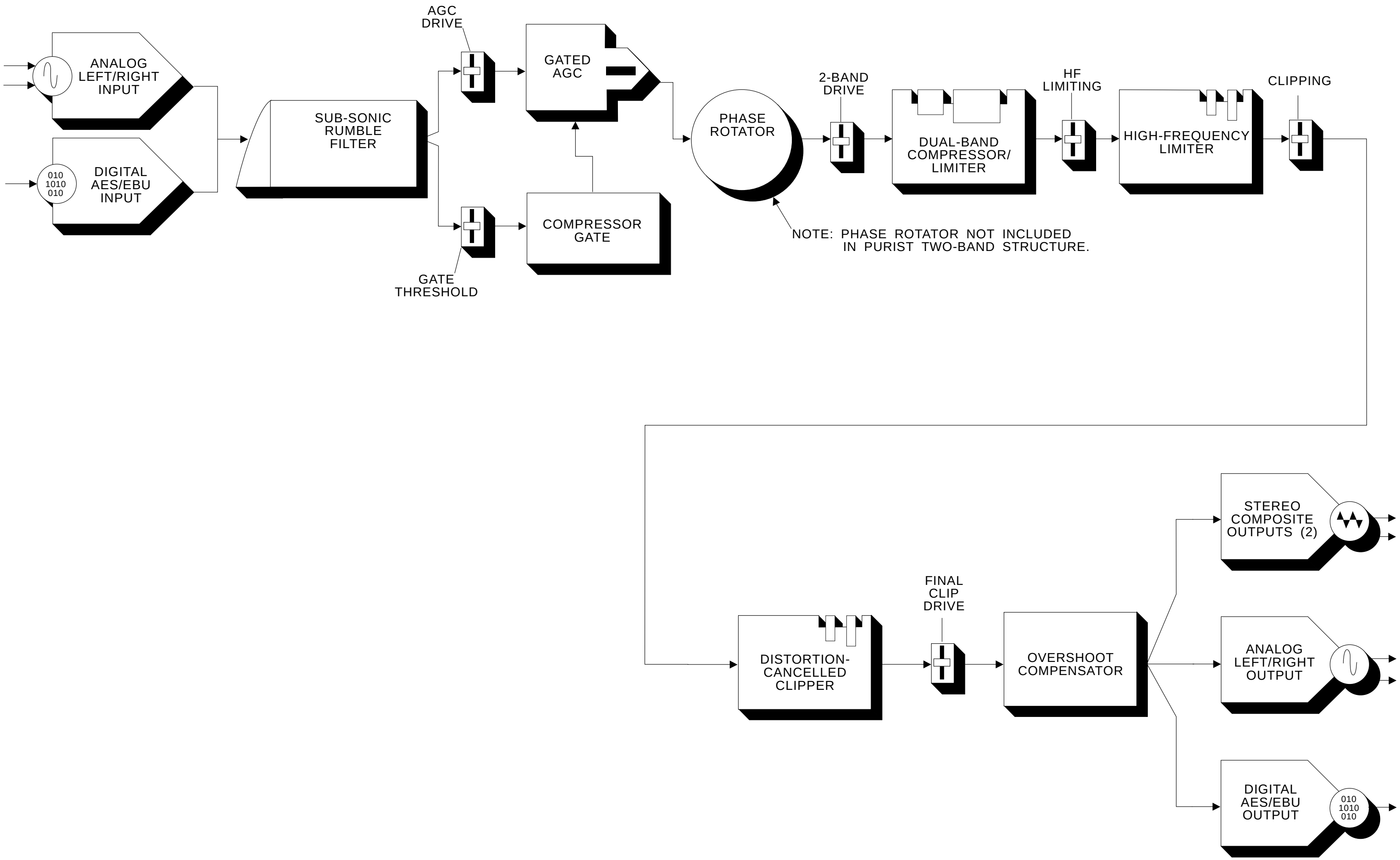
Both Two-Band Structures have an open, easy-to-listen-to sound that is similar to the source material if the source material is of good quality. However, if the spectral balance between the bass and high frequency energy of the program material is incorrect, the Two-Band Structures (when their BASS COUPLING controls are operated toward 0%) can gently correct it without introducing obvious coloration.

The Two-Band Structures are suitable for many easy-listening and adult contemporary formats whose programmers wish to maximize the average time spent listening by the average person in the audience. The Two-Band Structures' natural sound achieves very low listening fatigue, making the station very listenable for hours on end. Simultaneously, with the Two-Band Normal Structure, the station will be loud enough so that it will never get *lost on the dial*, and it will be consistent from source to source. Either structure provides an unprocessed sound that sounds just right on both music and voice when listened to on small table radios, car radios, portables, or home hi-fi systems.

Choosing the Two-Band Normal or Purist Structures

To achieve what we believe to be the most favorable balance between loudness and distortion, the Two-Band Normal Structure has a time-dispersion filter that makes voice peaks more symmetrical, minimizing voice distortion. Most stations will prefer to use the Two-Band Normal Structure because it is capable of about 3dB higher loudness than the Two-Band Purist Structure without significant loss of sound quality.

However, if you demand constant group delay at all frequencies, use the Two-Band Purist Structure instead, but be prepared to sacrifice considerable loudness as a result. Unfortunately, waveform fidelity must be achieved at the expense of using your peak modulation capability inefficiently — this is a limitation caused by the laws of physics. So be sure that you can *really* hear any degradation caused by frequency-dependent group delay before you make this significant trade-off. We believe that the audible effects of the frequency-dependent group delay in the Two-Band Structure are very subtle, at best, and are unlikely to be detected (let alone found objectionable) by your audience.



Two-Band Structure
(Simplified Block Diagram)

You can achieve higher loudness from the Two-Band Purist Structure if you process *all* voice material (live microphone and recorded) through an external time-dispersion filter prior to its application to the 8200's input. This way, you can maintain maximum fidelity on music, yet you can usually operate the CLIPPING control at –1 or 0. However, if certain asymmetrical voices are applied to the system unprocessed, you may find that they become highly distorted unless you reduce the CLIPPING control to –3 or below. This will cause a very noticeable loss of loudness on the air.

Setting Up the Two-Band Structure

To set up the Two-Band Structure, recall factory preset BB (Purist Classical) for a "smooth," unprocessed-sounding quality, or BA (Purist Processed) or CA (Normal Processed) for a louder, more processed sound. Of course, you can modify any preset using the LESS-MORE control or using the `FULL CONTROL` screen, and you can then store the resulting modified preset as a user preset for future recall at any time.

If you have the Multi-Band Structure, note that choosing the factory Multi-Band SLOW preset (DA) will almost always create an even more attractive, smooth, and consistent sound on pop music than will the Two-Band Structure. Additionally, because each band in the Multi-Band Structure handles a smaller part of the audio spectrum than each band in the Two-Band Structure, the Multi-Band Structure will create less spectral gain intermodulation when driven heavily. If you have the Multi-Band Structure, we therefore recommend using the Two-Band Structure only for classical or light classical music, or if your goal is to broadcast a sound that is more faithful to the frequency balances of the original program material than the sound produced by the slow Multi-Band Structure.

The Two-Band Factory Programming Presets

Only in units shipped without the Multi-Band Structure (see the inside of the front page for specific model numbers), there are fifteen **Two-Band** Factory Programming Presets whose names are specific formats. You are prompted to choose one of these at the end of the QUICK SETUP procedure. Except for CLASSICAL [which is based on BB (Purist Classical)], these presets are all derived from CA (the Normal Processed Structure), and differ only in the setting of the LESS-MORE control in preset CA. The Two-Band Factory Programming Presets are designed to help you get on the air quickly before you have had time to acquaint yourself with the processor. After you have become familiar with the processor, you will almost certainly want to experiment with the LESS-MORE control to fine-tune the processing to your taste.

Gain reduction metering.

Unlike the metering on some processors, when any OPTIMOD-FM gain reduction meter indicates full-scale (at its bottom), it means that its associated compressor has run out of gain reduction range, that the circuitry is being overloaded, and that various nastinesses are likely to commence.

Because the compressor in the Two-Band Structure has 25dB of gain reduction range, the meter should never come close to 25dB gain reduction if OPTIMOD-FM has been set up for a sane amount of gain reduction under ordinary program conditions.

But be aware of the different peak factors on voice and music — if voice and music are peaked identically on a VU meter, voice may cause up to 10dB more peak gain reduction than does music! (A PPM will indicate relative peak levels much more accurately.)

Using the Two-Band Structure for Classical Music

Classical music is traditionally broadcast with a wide dynamic range. However, with many recordings and live performances, the dynamic range is so great that the quiet passages disappear into the noise on most car, portable, and table radios. As a result, the listener either hears nothing, or must turn up the volume control to hear all the music. Then, when the music gets loud, the radio blasts and distorts, making the listening rather unpleasant.

The Two-Band Structure is well-suited for classical formats during daytime hours when most people in the audience are likely to be listening in autos or to be using the station for background music. This audience is best served when the dynamic range of the program material is compressed 10-15dB so that quiet passages in the music never fade into inaudibility under these less-favorable listening conditions. OPTIMOD-FM controls the level of the music in ways that are, for all practical purposes, inaudible to the listener. Low-level passages are increased in level by up to 10dB, while the dynamics of crescendos are maintained.

The same station may wish to switch to the Protection Structure during the evening hours when the audience is more likely to listen critically.

The Two-Band Structure’s Full Setup Controls

Each Two-Band factory preset has a LESS-MORE control that adjusts on-air loudness. LESS-MORE simultaneously adjusts all of the processing controls to optimize the trade-offs between unwanted side-effects as processing levels are decreased or increased.

If you wish, you may adjust the full controls to your own taste. Always start with LESS-MORE to get as close to your desired sound as possible. Then edit the full controls using the `FULL CONTROL` screen, and save those edits to a USER PRESET.

The controls available on the FULL CONTROL screen are:

AGC DRIVE control adjusts signal level going into the slow AGC, and therefore determines the amount of gain reduction in the AGC. This also adjusts the "idle gain" — the amount of gain reduction in the AGC section when the structure is gated. (It gates whenever the input level to the structure is below the threshold of gating.)

The total amount of gain reduction in the Two-Band Structure is the sum of the gain reduction in the AGC and the gain reduction in the two-band compressor. The total gain reduction determines how much the loudness of quiet passages will be increased (and, therefore, how consistent overall loudness will be). It is determined by the setting of the AGC DRIVE control, by the level at which the console VU meter or PPM is peaked, and by the setting of the 2B DRIVE Compressor control, discussed directly below.

2B DRIVE control adjusts signal level going into the two-band compressor, and therefore controls the density of output audio by determining the amount of gain reduction in the two-band compressor. The resulting sound texture can be open and transparent, solid and dense, or somewhere in between. The range is 0-25dB.

Regardless of the release time setting, we feel that the optimal amount of gain reduction in the two-band compressor for popular music and talk formats is 10-15dB. If less gain reduction is used, loudness can be lost. For classical formats, operating with 0-10dB of gain reduction (with the gain-riding AGC defeated), maintains a sense of dynamic range while still controlling levels effectively. Because OPTIMOD-FM's density gently increases between 0 and 10dB of compression, 10dB of compression sounds very natural, even on classical music.

The **RELEASE dB/SEC control** determines how fast the two-band compressor releases (and therefore how fast loudness increases) when the level of the program material decreases. It can be adjusted from 1dB/second (slow) to 20dB/second (fast). Settings toward 20dB/second result in a more consistently loud output, while settings toward 1dB/second allow a wider variation of dynamic range. The actual release time of the compressor is determined by both the setting of the RELEASE dB/SEC control and the dynamics and level of the program material. In general, you should use faster release times for mass-appeal pop or rock formats oriented toward younger audiences, and slower release times for more conservative, adult-oriented formats (particularly if women are an important part of your target audience).

The action of the RELEASE dB/SEC control has been optimized for resolution and adjustability. But its setting is *critical to sound quality* — listen carefully as you adjust it. There is a point beyond which increasing density (with faster settings of the RELEASE dB/SEC control) will no longer yield more loudness, and will simply degrade the punch and definition of the sound.

When the RELEASE dB/SEC control is set between 8 and 1dB/second (the slowest settings), the amount of gain reduction is surprisingly non-critical. Since gating prevents noise from being brought up during short pauses, and pumping does not occur at high levels of gain reduction, the primary danger of using large amounts of gain reduction is that the level of quiet passages in input material with wide dynamic range may eventually be increased unnaturally. Therefore, when you operate the RELEASE dB/SEC control between 8 and 2dB/second, it may be wise to defeat the gain-riding AGC and to permit the two-band compressor to perform all of the gain-riding. This will prevent excessive reduction of dynamic range, and will produce the most natural sound achievable from the Two-Band Structures.

With faster RELEASE dB/SEC control settings (above 8dB/second), the sound will change substantially with the amount of gain reduction in the two-band compressor. This means that you should activate the gain-riding AGC to ensure that the two-band compressor is always being driven at the level that produces the amount of gain reduction desired. Decide on the basis of listening tests how much gain reduction gives you the density you want without a creating feeling of over-compression and fatigue.

> Release in the two-band compressor automatically becomes faster as more gain reduction is applied (up to about 10dB). This makes the program progressively denser, creating a sense of increasing loudness even though peaks are not actually increasing. If the gain-riding AGC is defeated (with the AGC ON/OFF control), you can use this characteristic to preserve some feeling of dynamic range. Once 10dB of gain reduction is exceeded, full loudness is achieved — no further increase in short-term density occurs as more gain reduction is applied. This avoids the unnatural, fatiguing sound often produced by processors at high gain reduction levels, and makes OPTIMOD-FM remarkably resistant to operator gain-riding errors.

CLIPPING dB control adjusts signal level going into the distortion-canceling clippers and therefore determines the amount of peak limiting done by clipping. Range is −4dB to +2dB. This control and the FINAL CLIP DRIVE control govern the trade-off between loudness and distortion.

OPTIMOD-FM controls fast peaks by distortion-canceled clipping. The CLIPPING control adjusts the level of the audio driving the distortion-canceled clippers, and therefore adjusts the peak-to-average ratio. The loudness/distortion trade-off is primarily determined by the CLIPPING control.

Turning up the CLIPPING control drives the distortion-canceled clippers harder, reducing the peak-to-average ratio, and increasing the loudness on the air. When the amount of clipping is increased, the audible distortion caused by clipping is increased. Lower settings reduce loudness, of course, but result in a cleaner sound and better high-frequency response.

In our opinion, when the RELEASE dB/SEC control is set between 1 and 8dB/second, the best setting for the CLIPPING control is between −1 and 0. If the program material is clean, this setting produces an output that sounds undistorted even on high-quality receivers.

If you use faster settings of the RELEASE dB/SEC control, or if program material is not always clean, use lower settings of the CLIPPING control. Ultimately, your ears must judge how much distortion is acceptable. But audition difficult program material like live voice and piano before you make your final decision.

BASS COUPLING % control is used to set the balance between bass and the rest of the frequency spectrum.

The two-band compressor processes audio in a master band for all audio above approximately 200Hz, and a bass band for audio below approximately 200Hz. The BASS COUPLING control determines how closely the on-air balance of material below 200Hz matches that of the program material above 200Hz.

Settings toward 100% (wideband) make the output sound most like the input. Because setting the BASS COUPLING control at 100% will sometimes cause bass loss, the most accurate frequency balance will often be obtained with this control between 70% and 90%. The optimal setting depends on the amount of gain reduction applied. Adjust the BASS COUPLING control until the BASS G/R and MASTER G/R meters track as closely as possible.

With the two-band RELEASE dB/SEC control set to 2dB/second, setting the BASS COUPLING control toward 0% (independent) will produce a sound that is very open, natural, and non-fatiguing, even with large amounts of gain reduction. Such settings will provide a bass boost on some program material that lacks bass.

With fast release times, settings of the BASS COUPLING toward 100% (wideband) do not sound good. Instead, set the BASS COUPLING control toward 0% (independent). This combination of fast release and independent operation of the bands provides the maximum loudness and density on small radios achievable by the Two-Band Structure. But such processing may fatigue listeners with high-quality receivers, and also requires you to activate the AGC to control the average drive level into the two-band compressor, preventing uncontrolled build-up of program density.

HF LIM SOFT→HARD control determines how the processor avoids high-frequency overloads due to the pre-emphasis curve. When set toward −4dB (soft), the highs are controlled mostly by limiting (a form of dynamic filtering), which tends to soften highs — this could improve the sound of marginally distorted program material. When set toward +2dB (hard), the highs are controlled mostly by clipping, which could potentially distort highs.

Control of highs by limiting tends to slightly dull the sound. Control of highs by clipping doesn't reduce brightness, but the resulting sound can tend towards grittiness and smearing.

Because the OPTIMOD-FM distortion-canceling clipper does not produce significant distortion at low frequencies, the HF LIM SOFT→HARD control will have a different effect on clipping distortion than you might expect. Outright break-up (principally sibilance splatter) will not occur — you must listen to the upper midrange and the highs to hear the effect of the clipper. Program material containing highly equalized hi-hat cymbals will clearly demonstrate the effect of adjusting the control.

When the CLIPPING control is set to 0dB or below and the RELEASE dB/SEC control is set slower than 8dB/second, it is possible to set the HF LIM SOFT→HARD control to +2dB without producing objectionable distortion (provided that the program material is very clean). If the CLIPPING control is set above 0 and/or faster release times are used (such that greater level and density is produced), it is usually necessary to readjust the HF LIM SOFT→HARD control closer to −2dB (soft) to avoid objectionable distortion. Fortunately, the high-frequency limiter *knows* that greater density and level have been produced when these other controls are set this way, and most of the necessary increases in high-frequency limiting will occur automatically. In fact, you will clearly hear a loss of highs when you adjust any control to produce greater loudness and density — this is an automatic response to the loudness/brightness/distortion trade-off inherent to all broadcast processing.

FINAL CLIP DRIVE control determines the level driving the final clipper that performs protection peak limiting. (This clipper follows the distortion-canceling clipper system, and is not itself distortion-canceling.) The adjustment of this control is *very critical* — changes of 0.1dB make clearly audible differences in the amount of distortion produced by the processing. In most cases, we recommend that the user not adjust this control and use the factory preset settings instead; the control has only been made available for experienced, sophisticated users who need to achieve the absolute maximum on-air loudness and who are willing to take the time necessary to listen to many different kinds of program material to verify that nothing falls apart after the clipper drive has been increased. The effect of adjusting this control is very similar to the effect of changing the amount of clipping in a composite clipper, except that in the 8200 (unlike a composite clipper), the SCA region of the baseband spectrum is always perfectly protected from interference.

GATE THRESHOLD dB control determines the lowest input level that will be recognized as program by OPTIMOD-FM; lower levels are considered to be noise or background sounds, and cause the compressor to gate, effectively freezing its gain.

The two-band gain reduction will eventually recover to 0dB and the AGC gain reduction will eventually recover to –10dB even when the compressor gate is gated. However, recovery is slow enough to be imperceptible. This avoids OPTIMOD-FM's getting stuck with a large amount of gain reduction on a long, low-level musical passage immediately following a loud passage.

It is common to set the GATE THRESHOLD control to –40. Higher settings are primarily useful for radio drama, outside sports broadcasts, and other non-musical programming which contain ambiance, low-level crowd noise, and the like. Slightly higher settings may increase the musicality of the compression by slowing down recovery on moderate-level to low-level musical passages. When such passages cause the gate to cycle on and off, recovery time will be slowed down by the ratio of the "on time" to the "off time." This effectively slows down the release time as the input gets quieter and quieter, thus preserving musical values in material with wide dynamic range (classical music, for example).

AGC ON/OFF control activates or defeats the slow AGC prior to the two-band compressor.

30Hz HPF control determines if the 30Hz high-pass filter prior to the AGC is in or out of the signal path.

The 30Hz high-pass filter has an 18dB/octave slope, is down 0.5dB at 30Hz, and is placed before the gain-riding AGC. It can be placed in or out-of-circuit by the 30Hz HPF control.

This filter eliminates modulation-wasting subsonic energy from acoustic and turntable rumble, and removes most of the energy from *pops* caused by breath blasts into microphones. It prevents any such subsonic energy from modulating the audio processor's AGC and compressor control signals (which could cause unpleasant distortion), and prevents the automatic frequency control loops in FM exciters from introducing modulation distortion into the audio or even becoming entirely unlocked.

The cutoff frequency of this filter is so low that the only common musical instruments producing lower fundamental frequencies are the pipe organ and synthesizer. The bass energy in most pop music occurs above 40Hz. The ringing introduced by the filter is insignificant. The ear is very insensitive to ringing in this frequency range. To put the issue in perspective, the ringing is comparable to that introduced by a well-designed vented box loudspeaker with a 30Hz cutoff. For all of these reasons, we recommend operating the system with the 30Hz HPF control set ON. We can only recommend setting it OFF if the programming is exclusively from compact disc or tape mastered from non-vinyl sources, and if high-pass filters are applied to all live microphones to prevent pops or rumble from disturbing later components in the broadcast chain.

If you are using the Two-Band Purist Structure, be aware that the high-pass filter does not have constant group delay at all frequencies. However, for reasons discussed above, we still recommend its use.

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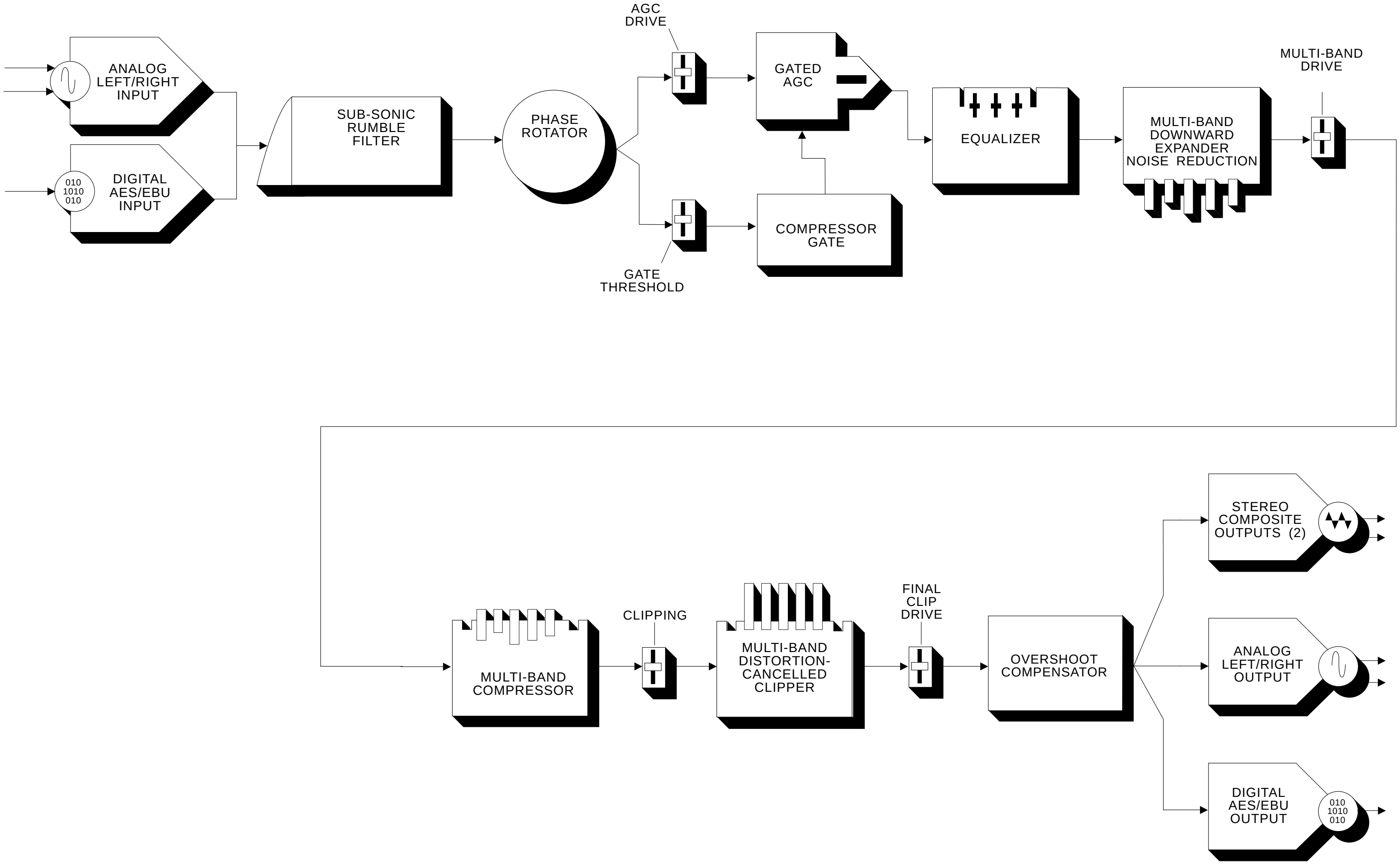
The Multi-Band Structure

The Multi-Band Structure consists of a slow gain-riding AGC, a four-band equalizer, a five-band compressor, a dynamic single-ended noise reduction system, a multi-band distortion-cancelled clipper and an overshoot compensator.

Unlike the Two-Band Structure, whose two-band compressor has a continuously-variable release time, the release time of the multi-band compressor is switchable to four settings: SLOW, MSLOW (medium-slow), MFAST (medium-fast), and FAST. Each setting makes a very significant difference in the overall flavor and quality of the sound.

When the input is noisy, you can sometimes reduce the noise by activating the **single-ended noise reduction system**. Functionally, the single-ended noise reduction system combines a broadband downward expander with a program-dependent low-pass filter. This noise reduction can be valuable in reducing audible hiss, rumble, or ambient studio noise on-air.



Multi-Band Structure
(Simplified Block Diagram)

Putting the Multi-Band Structure on the Air

The Multi-Band Structure is very flexible, enabling you to fine-tune your on-air sound to complement your programming. There are four basic Factory Presets for the Multi-Band Structure:

##	PRESET NAME/RELEASE TIME
DA	5B-SLOW
DB	5B-MEDIUM SLOW
DC	5B-MEDIUM FAST
DD	5B-FAST

Each of these presets can be edited with the LESS-MORE control to optimize the tradeoff between loudness, brightness, and distortion according to the needs of the format.

5B-SLOW produces a very punchy, clean, open sound that is ideal for Adult Contemporary, Soft Rock, Soft Urban, New Age, and other adult-oriented formats whose success depends on attracting and holding audiences for very long periods of time. SLOW produces an *unprocessed* sound with a nice sense of dynamic range. With these settings, the Multi-Band Structure provides gentle automatic equalization to keep the frequency balance consistent from record to record (especially those recorded in different eras). And for background music formats, these settings ensure that your sound doesn't lose its highs and lows. Because it creates a more consistent frequency balance between different pieces of source material than does the Two-Band Structure, SLOW is almost always preferable to the Two-Band Structure for any popular music format.

5B-MEDIUM SLOW is appropriate for more adult-oriented formats that need a glossy show-business sound, yet whose ratings depend on maintaining a longer time spent listening than do conventional CHR formats. With the single-ended noise reduction activated, it is also appropriate for Talk and News formats. This is the sound texture for the station that values a clean, easy-to-listen-to sound with a tasteful amount of punch, presence, and brightness added when appropriate. This is an *unprocessed* sound that sounds just right on music and voice when listened to on small table radios, car radios, portables, or home hi-fi systems.

5B-MEDIUM FAST is appropriate for the same formats as FAST, but where the market is not engaged in loudness wars. MEDIUM FAST still provides a high-energy sound, but with less density (and therefore less danger of fatiguing listeners) than FAST. In talk formats, processing for this sound keeps the levels of announcers and guests consistent, pulls low-grade phone calls out of the mud, and keeps a proper balance between voice and commercials. Voice is the most difficult audio to process, but these settings result in a favorable trade-off between consistency, presence, and distortion.

5B-FAST produces a very loud, consistent, dramatic, and rather synthetic sound that is ideal for a highly competitive Contemporary Hit Radio (CHR) format whose ratings depend on attracting a large number of listeners (high "cume") but which does not assume that a listener will listen to the station for hours at a time. This is the major market *competitive sound*, emphasizing loudness as well as clean audio. The sound from cut to cut and announcer to announcer is remarkably consistent as the texture of music is noticeably altered to a standard. Bass has an ever-present punch, there is always a sense of presence, and highs are in perfect balance to the mids, no matter what was on the original recording.

The factory presets for this sound are quite different than for the other three release time settings. The amount of gain reduction in the multi-band compressor is substantially lower (so that it operates more like a limiter than like a compressor), and the release time of the gain-riding AGC is speeded up (so that it provides compression and some increase of density). We made these tradeoffs to prevent excessive build-up of density.

Even with these settings, your sound is getting farther away from the balance and texture of the original recording. We think that this is as far as processing can go without causing unacceptable listener fatigue. However, this sound may be quite useful for stations that are ordinarily heard very softly in the background because it improves intelligibility under these quiet listening conditions. Stations that are ordinarily played louder will probably prefer one of the slower release times, where the multi-band compressor takes more gain reduction and where the AGC is operated slowly for gentle gain riding only. These slower sounds are less consistent than those produced by the FAST setting. SLOW preserves more of the source's frequency balance, and will be less dense and less fatiguing when the radio is played loudly.

In addition to the four basic factory presets, there are fifteen Factory Programming Presets whose names are specific formats. You are prompted to choose one of these at the end of the QUICK SETUP procedure. Except for CLASSICAL, all these presets are derived from the four basic factory presets for the Multi-Band Structure: DA for SLOW release time, DB for MEDIUM SLOW release time, DC for MEDIUM FAST release time, and DD for FAST release time. (See the discussion of AGC RELEASE dB/SEC on page 3-35 below.)

Start with one of these presets. Spend some time listening critically to your on-air sound. Listen to a wide range of program material typical of your format, and listen on several types of radios (not just on your studio monitors). Then, if you wish, customize your sound using the information in "Customizing the Settings," which follows.

Customizing The Settings

The controls in the Multi-Band Structure give you the flexibility to customize your station sound. But as with any audio processing system, proper adjustment of these controls requires proper balancing of the trade-offs between loudness, density, brightness, and audible distortion. The following provides the information you need to adjust the Multi-Band Structure controls to suit your format, taste, and competitive situation.

The Multi-Band Structure's Full Setup Controls

Each Multi-Band factory preset has a LESS-MORE control that adjusts on-air loudness by altering the amount of processing. LESS-MORE simultaneously adjusts all of the processing controls to optimizing the trade-offs between unwanted side-effects.

If you wish, you may adjust the full controls to your own taste. Always start with LESS-MORE to get as close to your desired sound as possible. Then edit the full controls using the FULL CONTROL screen, and save those edits to a USER PRESET.

The controls available on the FULL CONTROL screen are:

AGC DRIVE control adjusts signal level going into the slow AGC, and therefore determines the amount of gain reduction in the AGC. This also adjusts the "idle gain" — the amount of gain reduction in the AGC section when the structure is gated. (It gates whenever the input level to the structure is below the threshold of gating.)

The total amount of gain reduction in the Multi-Band Structure is the sum of the gain reduction in the AGC and the gain reduction in the multi-band compressor. The total system gain reduction determines how much the loudness of quiet passages will be increased (and, therefore, how consistent overall loudness will be). It is determined by the setting of the AGC DRIVE control, by the level at which the console VU meter or PPM is peaked, and by the setting of the MULTI-BAND DRIVE (compressor) control.

The **AGC RELEASE dB/SEC control** can be adjusted from 0.5dB/second (slow) to 20dB/second (fast). The increase in density caused by setting the AGC RELEASE dB/SEC control FAST sounds different than the increase in density caused by setting the MULTI-BAND RELEASE control FAST, and you can trade the two off to produce different effects.

Unless it is purposely speeded-up (with the AGC RELEASE dB/SEC control), the automatic gain control (AGC) that occurs in the AGC prior to the multi-band compressor makes audio levels more consistent without significantly altering texture. Then the multi-band compression and associated multi-band clipper audibly change the density of the sound and dynamically re-equalize it as necessary (booming bass is tightened; weak, thin bass is brought up; highs are always present and consistent in level).

The various combinations of *AGC* and *compression* offer great flexibility:

Light AGC + light compression yields a wide sense of dynamics, with a small amount of automatic re-equalization.

Moderate AGC + light compression produces an open, natural quality with automatic re-equalization and increased consistency of frequency balance.

Moderate AGC + moderate compression gives a more *dense* sound, particularly as the release time of the multi-band compressor is sped up.

Moderate AGC + heavy compression (particularly with FAST multi-band release time) results in a "wall of sound" effect which may cause listener fatigue.

Adjust the AGC (with the AGC DRIVE control) to produce the desired amount of AGC action, and then fine-tune the compression and clipping with the Multi-Band Structure's controls.

MULTI-BAND DRIVE control adjusts signal level going into the multi-band compressor, and therefore determines the average amount of gain reduction in the multi-band compressor. Range is 25dB.

Adjust the MULTI-BAND DRIVE control to your taste and format requirements. Used lightly with SLOW or MSLOW (medium-slow) release time, the multi-band compressor produces an open, re-equalized sound. The multi-band compressor can increase audio density when operated at FAST or MFAST (medium-fast) release because it acts more and more like a fast limiter (not a compressor) as the release time is shortened. With FAST and MFAST (medium-fast) release times, density also increases when you increase the drive level into the multi-band compressor because these faster release times produce more limiting action. Increasing density can make sounds seem louder, but can also result in an unattractive *busier, flatter,* or *denser* sound. It is very important to be aware of the many negative subjective side effects of excessive density when setting controls that affect the density of the processed sound.

The MULTI-BAND DRIVE interacts with the MULTI-BAND RELEASE. With slower release time settings, increasing the MULTI-BAND DRIVE control scarcely affects density. Instead, the primary danger is that the excessive drive will cause noise to be excessively increased when the program material becomes quiet.

> You can minimize this effect by carefully setting the GATE THRESHOLD control to "freeze" the gain when the input gets quiet and/or by activating the single-ended noise reduction.

When the release time of the multi-band compressor is set to FAST, or MEDIUM FAST (presets DD and DC), the setting of the MULTI-BAND DRIVE control becomes much more critical to sound quality because density increases as the control is turned up. Listen carefully as you adjust it. With these fast release times, there is a point beyond which increasing multi-band compressor drive will no longer yield more loudness, and will simply degrade the punch and definition of the sound.

We recommend no more than 10dB gain reduction as shown on the meters for Band 3. More than 10dB, particularly with the FAST release time, will often create a "wall of sound" effect that many find fatiguing.

> To avoid excessive density with FAST multi-band release time, we recommend using no more than 5dB gain reduction in Band 3, and compensating for any lost loudness by speeding up the AGC RELEASE dB/SEC instead. This is what we did in the factory LESS-MORE presets for the FAST multi-band release time.

The **MULTI-BAND RELEASE control** can be switched to FAST, MFAST (medium-fast), MSLOW (medium-slow), and SLOW. See the detailed description on page 3-33.

MULTI-BAND CLIPping control adjusts signal level going into the multi-band clippers and therefore determines the amount of peak limiting done by clipping. Range is –4dB to +5dB. This control and the FINAL CLIP DRIVE control govern the trade-off between loudness and distortion. Note that these ranges are relative, and do not indicate the exact amount of clipping. 0dB refers to the average setting that we have found to be the best compromise for many settings of the other controls.

If the MULTI-BAND RELEASE control is set to FAST or MFAST (medium-fast), perceived clipping distortion will increase as the MULTI-BAND DRIVE control is advanced, and the MULTI-BAND CLIP control may have to be turned down to compensate. To best understand how to make loudness/distortion trade-offs, perhaps the wisest thing to do is to recall a given preset (DA, DB, DC, or DD), and then to adjust the LESS-MORE control to several settings throughout its range. At each setting of the LESS-MORE control, examine the settings of the MULTI-BAND DRIVE control, the MULTI-BAND CLIP control, and the FINAL CLIPPING DRIVE control. (You can see them by calling up the FULL CONTROL screen by pressing the FULL CONTROL soft key.) This way, you can see how the factory programmers made the trade-offs between the settings of the various distortion-determining controls at various levels of processing.

LOW BASS BOOST control, the Multi-Band Structure's low bass equalization control, is designed to add *punch* and *slam* to rock and urban music. It provides a shelving boost in two ranges with two switchable characteristics. 2P provides a range of 0dB to +12dB (in 2dB steps) with a 12dB/octave slope. 3P provides a range of 0dB to +12dB (in 2dB steps) at 110Hz and below with an 18dB/octave slope.

Because the Multi-Band Structure often increases the brightness of program material, some bass boost is usually desirable to keep the sound spectrally well-balanced. Adjustment of bass equalization must be determined by individual taste and by the requirements of your format. Be sure to listen on a wide variety of radios — it is possible to create severe distortion on poor quality speakers by over-equalizing the bass. Be careful!

The moderate-slope (12dB/octave) shelving boost achieves a bass boost that is more audible on smaller radios, but which can sound boomier on high-quality receivers. The steep-slope (18dB/octave) shelving boost creates a solid, punchy bass from the better consumer radios with decent bass response. There are no easy choices here; you must choose the characteristic you want by identifying your target audience and the receivers they are most likely to be using. Regardless of which curve you use, we recommend a +2 to +4dB boost for most formats.

> The LOW BASS BOOST control steps first through the 2P curves and then through the 3P curves. For example, a 4dB boost with a 12dB/octave slope would appear as "2P +4," and a 10dB boost with an 18dB/octave slope would appear as "3P +10."

MID BASS BOOST control provides a 12dB/octave shelving boost at 200Hz. Use it in conjunction with the LOW BASS BOOST control to tailor your on-air bass to your exact requirements. A mid-bass boost is mainly useful to stations that program to an audience likely to be listening on smaller radios. It can force a thin-sounding radio with a small speaker to seem to have more bass. However, bass boost in this frequency range can make larger radios sound very muddy and boomy, so adjust the MID BASS BOOST control with great care, listening to both small radios and radios with good bass response.

PRESENCE dB control is a 6dB/octave peaking equalizer centered at 3kHz that affects several bands in the multi-band compressor.

The audible effect of the PRESENCE control is closely associated with the amount of gain reduction in the 3.7kHz band. With small amounts of gain reduction, the effect is an actual boost in the amount of power in the presence region, which can increase the loudness of such material substantially. As you increase the gain reduction in the 3.7kHz band (by turning the

MULTI-BAND DRIVE control up), the PRESENCE control will have progressively less audible effect. The compressor for the 3.7kHz band will tend to reduce the effect of the PRESENCE boost (in an attempt to keep the gain constant) to prevent excessive stridency in program material that already has a great deal of presence power. So with large amounts of gain reduction, the density of presence region energy will be increased more than will the level of energy in that region. Because the 3.7kHz band compressor governs the gain reduction in the 6.2kHz band also, turning up the PRESENCE control will *decrease* energy in the 6.2kHz band, since you will be increasing the gain reduction in *both* the 3.7kHz and 6.2kHz bands. You may wish to compensate for this effect by turning up the BRILLIANCE control.

Nevertheless, if less than 5dB gain reduction is used in the 3.7kHz and 6.2kHz bands, the PRESENCE control will have a significant audible effect almost all of the time. Use the PRESENCE control with caution. Excessive presence boost tends to be audibly strident and fatiguing. And the sound quality, although loud, can be very irritating. We suggest a maximum of 2–3dB boost, although 6dB can be achieved.

BRILLIANCE dB control increases the gain (prior to limiting) of Band 5 (6.2kHz and above). It produces an audibly attractive effect with increased *air* and *transparency* (similar to that of a psychoacoustic exciter). Unlike the PRESENCE control, the BRILLIANCE control will have an audible effect at all times.

Excessive brilliance boost can exaggerate tape hiss and distortion in program material that is less than perfectly clean. We suggest no more than 4dB boost as a practical maximum, unless source material is primarily from compact discs of recently-recorded material.

FINAL CLIP DRIVE control determines the level driving the final clipper that performs peak limiting. The adjustment of this control is very critical — changes of 0.1dB make clearly audible differences in the amount of distortion produced by the processing. In most cases, we recommend that the user not adjust this control and use the factory preset settings instead (as determined by a factory preset, or by a LESS-MORE setting); the control has only been made available for experienced, sophisticated users who need to achieve the absolute maximum on-air loudness and who are willing to take the time necessary to listen to many different kinds of program material to verify that nothing falls apart after the clipper drive has been increased. The effect of adjusting this control is very similar to the effect of changing the amount of clipping in a composite clipper, except that, in the 8200 (unlike a composite clipper), the SCA region of the baseband spectrum is always perfectly protected from interference.

GATE THRESHOLD dB control determines the lowest input level that will be recognized as program by OPTIMOD-FM; lower levels are considered to be noise or background sounds, and cause the AGC to gate, effectively freezing gain to prevent noise breathing.

The gate causes the gain reduction in bands 2 and 3 of the multi-band compressor to quickly move to the average gain reduction occurring in those bands when the gate first turned on. This prevents obvious midrange coloration under gated conditions, because bands 2 and 3 have the same gain.

The gate also independently freezes the gain of the two highest-frequency bands (forcing the gain of the highest-frequency band to be identical to its lower neighbor), and independently sets the gain of the lowest-frequency band according to the setting of the DJ BASS

BOOST control (see immediately below). Thus, without introducing blatant coloration, the gating smoothly preserves the average overall frequency response "tilt" of the multi-band compressor, broadly maintaining the "automatic equalization" curve it generates for a given piece of program material.

NOTE: If the GATE THRESHOLD control is turned OFF, the DJ BASS BOOST control is disabled.

HF CLIPPING dB control sets the threshold of clipping in bands 4 and 5 with reference to the overall threshold set by the MULTI-BAND CLIP control. The range is 0 to +6dB. We have made this control available for some major-market customers who prefer a brighter sound at the expense of audible distortion on a significant amount of program material. We recommend a setting of 0 for all program material. *This control is for audio processing experts only*; it is important to listen to a very wide range of program material before deciding to set this control above 0. When the control is set above 0, there will definitely be some program material that sounds harsh and distorted, and you must decide if the trade-off against brighter sound and more vocal presence is acceptable to your taste.

DJ BASS BOOST control determines the amount of bass boost produced on some male voices. (If the GATE THRESHOLD control is turned OFF, the DJ BASS BOOST setting is disabled.) In its default OFF position, it causes the gain reduction of the lowest-frequency band to move quickly to the same gain reduction as its nearest neighbor when gated. This fights any tendency of the lowest frequency band to develop significantly more gain than its neighbor when processing voice because voice will activate the gate frequently. Each time it does so, it will reset the gain of the lowest frequency band so that the gains of the two bottom bands are equal and the response in this frequency range is flat. The result is natural-sounding bass on male voice.

If you like a larger-than-life, "chesty" sound on male voice, set this control ON. When the control is ON, the processing simply freezes the gain of the lowest band under gated conditions. Accordingly, there can be a large average gain difference between the two low-frequency bands and the system can produce considerable dynamic bass boost on voice.

This will be highly dependent on the fundamental frequency of the voice. If the fundamental frequency is far above 100Hz there will be little voice energy in the bottom band and little or no audio bass boost can occur even if the gain of the bottom band is higher than the gain of its neighbor. As the fundamental frequency moves lower, more of this energy leaks into the bottom band, and you hear more bass boost. If the fundamental frequency is *very* low (a rarity), there will be enough energy in the bottom band to force significant gain reduction, and you will hear less bass boost than if the fundamental frequency were a bit higher.

BASS COUPLING % control determines the extent to which the gain of band 1 (below 100Hz) is determined by and follows the gain of band 2 (mid-bass). Set towards 100% (fully coupled) it reduces the amount of dynamic bass boost, preventing unnatural bass boost in light pop and talk formats. Set towards 0% (independent), it permits frequencies below 100Hz (the "slam" region) to have maximum impact in modern rock, urban, dance, rap, and other music where bass punch is crucial. The default setting is 30%.

DOWNward EXPANDer THReshold control determines the level below which the single-ended noise reduction system's downward expander begins to decrease system gain, and below which the high frequencies begin to become low-pass filtered to reduce perceived noise. Activate the single-ended dynamic noise reduction by setting the DOWN EXPAND THR control to a setting other than OFF.

The single-ended noise reduction system combines a broadband downward expander with a program-dependent low-pass filter. These functions are achieved by causing extra gain reduction in the multi-band compressor. You can see the effect of this extra gain reduction on the gain reduction meters.

Ordinarily, the gating on the AGC and multi-band limiter will prevent objectionable build-up of noise, and you will want to use the single-ended noise reduction only on unusually noisy program material. Modern commercial recordings will almost never need it. We expect that its main use will be in talk-oriented programming, including sports.

Please note that it is impossible to design such a system to handle all program material without audible side-effects. You will get best results if you set the DOWN EXPAND THR control of the noise reduction system to complement the program material you are processing. The DOWN EXPAND THR should be set higher when the input is noisy and lower when the input is relatively quiet. The best way to adjust the DOWN EXPAND THR control is to start with the control set very high. Reduce the setting of control while watching the gain reduction meters. Eventually, you will see the gain increase in sync with the program. Go further until you begin to hear *noise modulation* — a puffing or breathing sound (the input noise) in sync with the input program material. Set the DOWN EXPAND THR control higher until you can no longer hear the noise modulation. This is the best setting.

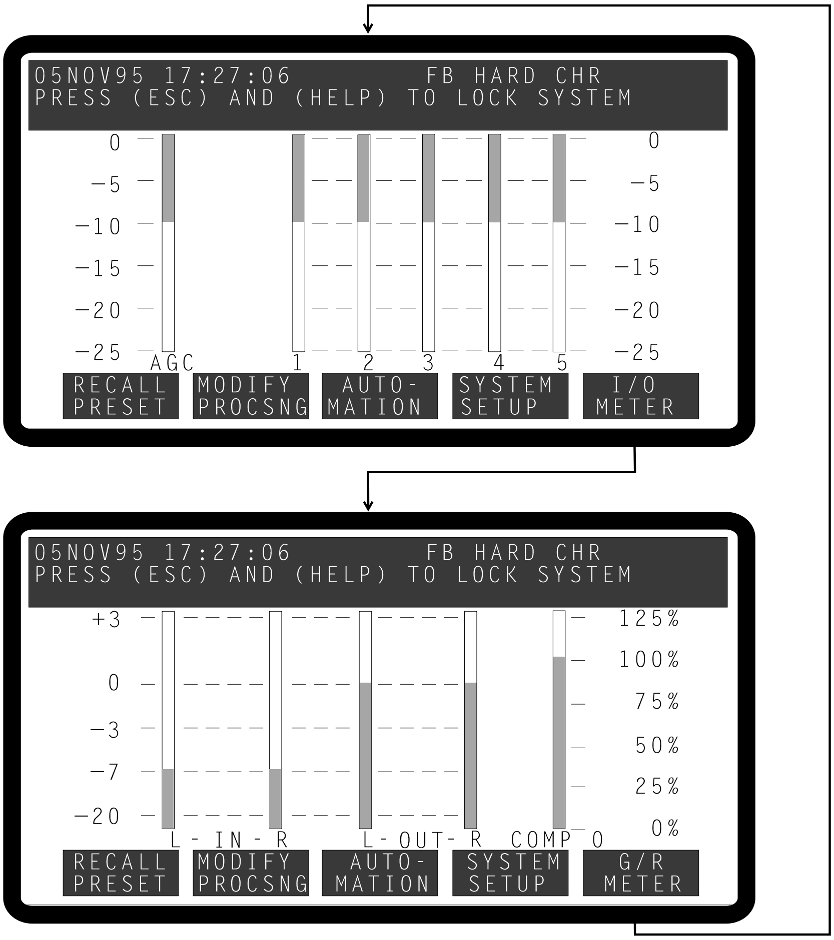
Obviously the correct setting will be different for a sporting event than for classical music. It may be wise to define several presets with different settings of the THRESHOLD control, and to recall the preset that complements the program material of the moment.

Note also that it is virtually impossible to achieve undetectable dynamic noise reduction of program material that is extremely noisy to begin with, because the noise is never masked by the program. It is probably wiser to defeat the dynamic noise reduction with this sort of material (traffic reports from helicopters and the like) to avoid objectionable side-effects. You must let your ears guide you.

30Hz HPF control determines if the 30Hz 18dB/octave high-pass filter prior to the AGC is in or out of the signal path. For most situations, we recommend that it be IN, as explained on page 3-15.

AGC ON/OFF control activates or defeats the slow AGC prior to the multi-band compressor.

IDLE G/R and IDLE I/O Screens



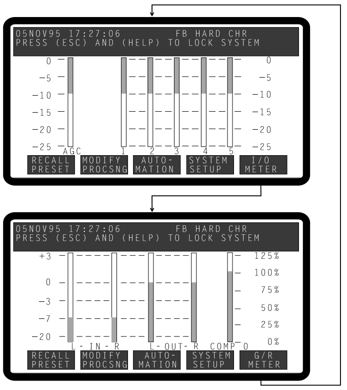
The `IDLE G/R` screen shows gain reduction meters for the processing structure in use.

The `IDLE I/O` screen shows left/right input/output levels, and if the stereo encoder is present, composite modulation in %.

Press the fifth soft key (I/O METER or G/R METER) to toggle between the two screens.

NOTE: Functions that are locked (other than RECALL PRESET) will not be shown. An ENTER PASCODE button is shown in place of the SYSTEM SETUP button, and can be used to enter in a valid pascode. (See `IDLE LOCKOUT` screens.)

IDLE G/R and IDLE I/O Screen Buttons



RECALL PRESET

Presets are stored settings of the processing controls (See Section 1). Press RECALL PRESET to see a list of all stored presets. You can select a preset from the list, and recall that preset on the air.

MODIFY PROCSNG

Press MODIFY PROCSNG to get to the controls that you can adjust to change the sound of the processing on the air.

AUTO-MATION

Automation is automatic programmed change of presets at scheduled times. This option is available with software version 1.0 and higher. Press AUTOMATION to see the list of programmed changes, and to change this list.

SYSTEM SETUP

Use SYSTEM SETUP to:

- Check software version number and hardware installed
- Run QUICK SETUP (for initial system setup).
- Calibrate Input/Output levels
- Set the time and date
- Program pascodes and remote control functions.

I/O METER

From `IDLE G/R` screen, press I/O METER to see Input/Output levels on the screen instead of gain reduction levels. Metering will show left and right input and output levels, and, if the stereo encoder is present, composite modulation in %.

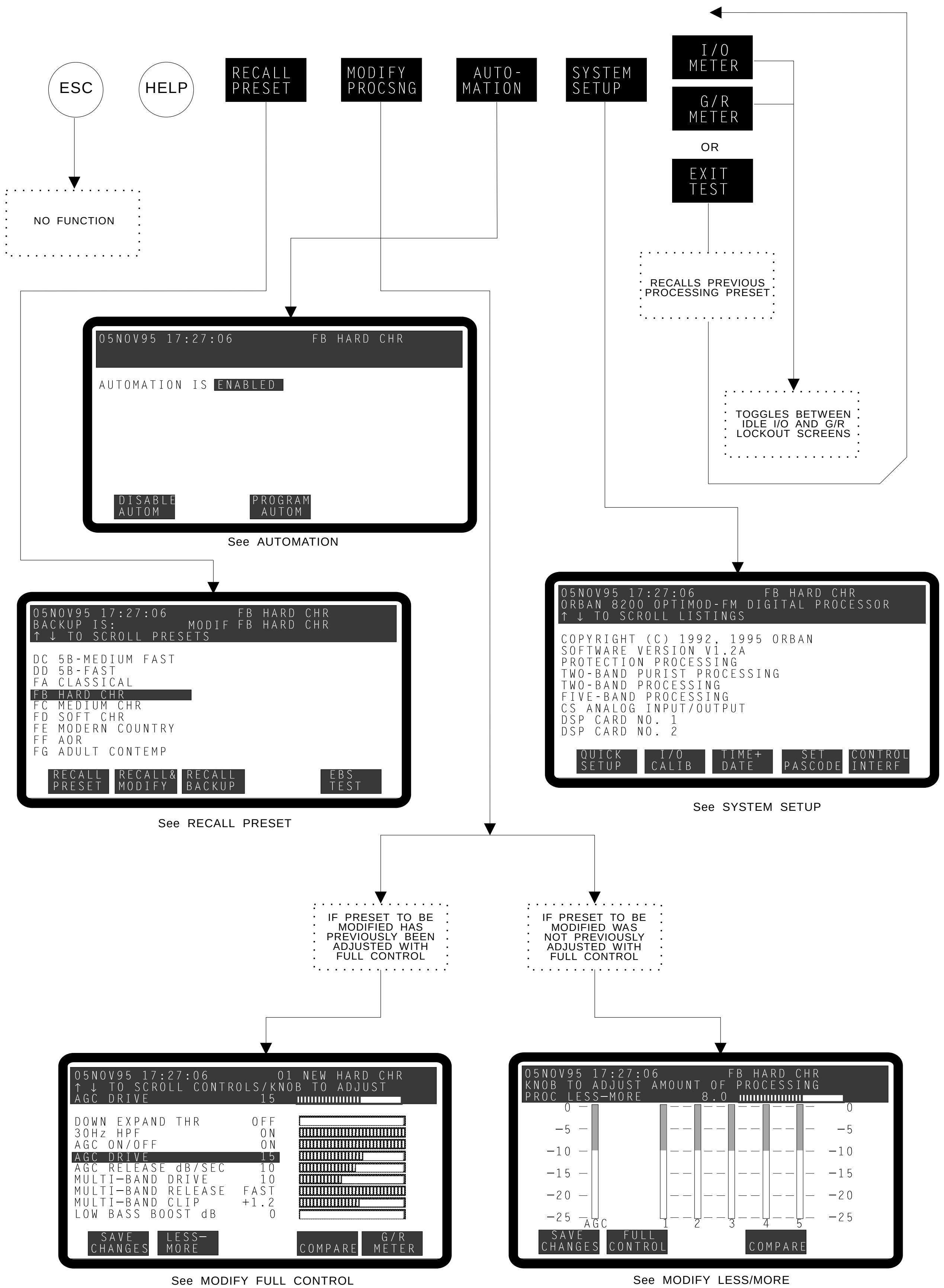
G/R METER

From `IDLE I/O` screen, press G/R METER to see gain reduction levels on the screen instead of Input/Output levels. Metering will show gain reduction for the processing structure in use. This button is not displayed if a special tone preset (e.g., TO, BY, EB,) is selected.

EXIT TEST

From `IDLE I/O LOCKOUT` screen, press EXIT TEST to exit the test preset on air and recall previous processing preset. This button is displayed if a special test preset is selected.

IDLE G/R and IDLE I/O Flow Chart



IDLE G/R and IDLE I/O Additional Screen Information

`IDLE G/R` screen: When a PC is connected to your 8200 for remote control operation, the second line of the `IDLE G/R` screen reads: PC ONLINE: Log-on name; the third line of the `IDLE G/R` screen reads: TO DISCONNECT, ENTER PASCODE.

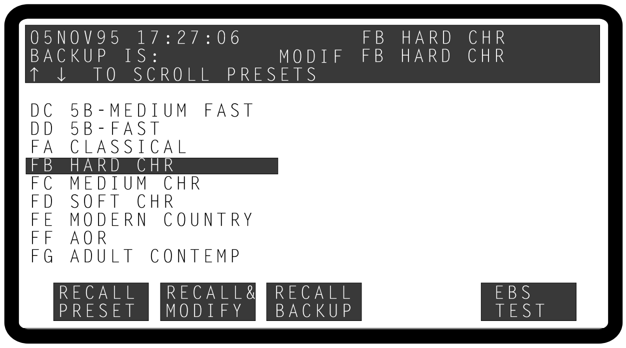
`IDLE I/O` screen: When Digital I/O is fitted, ANALOG IN, or DIGITAL IN appears on line 3 of the `IDLE I/O` screen, depending upon which is active. TX1, TX2, SCA1 and SCA appear at the end of the line if the function they represent is active (TX1 OSHOOT COMP, etc)

RECALL PRESET Screen

Preset List

User Preset	**Preset Name**
01 - 32	USER PRESETS
AA	PROTECTION 0dB
AB	PROTECTION 5dB
BA	2B-PUR PROCESSED
BB	2B-PUR CLASSICAL
CA	2B-NOR PROCESSED
DA	5B-SLOW
DB	5B-MEDIUM SLOW
DC	5B-MEDIUM FAST
DD	5B-FAST
FA	CLASSICAL
FB	HARD CHR
FC	MEDIUM CHR
FD	SOFT CHR
FE	MODERN COUNTRY
FF	AOR (Album-Oriented Rock)
FG	ADULT CONTEMP
FH	URBAN
FI	SOFT URBAN
FJ	NEW AGE
FK	OLDIES
FL	JAZZ
FM	BEAUTIFUL
FN	MOR (Middle-of-the-Road)
FO	TALK
BY	BYPASS
TO	USER TONE

RECALL PRESET Screen and Buttons



Presets are stored settings of the processing controls.
Use the ↑ and ↓ keys to select a preset from the list. Press RECALL PRESET to recall that preset on the air.

RECALL PRESET

Press RECALL PRESET to put the selected (highlighted) preset on the air. If the selected preset is the same as the preset already on the air, the sound of the station will not change.
When you press RECALL PRESET, the previous preset is remembered. To return to it, press RECALL BACKUP.

RECALL& MODIFY

Press RECALL& MODIFY to put the selected (highlighted) preset on the air, then get to the controls that you can adjust to change the sound of the processing on the air. This is the same as pressing RECALL PRESET, ESC, then MODIFY PROCSNG.

RECALL BACKUP

When you recall a preset, the preset that was on the air previously is remembered. Or when you modify a preset, the setting of the controls before you changed them are remembered. The number and name of the backup preset is shown on the screen on line 2.
Press RECALL BACKUP to recall the previous preset or settings.

EBS TEST

Press to access EBS TEST screen. This screen makes you verify you really want an EBS TEST on air, before continuing.

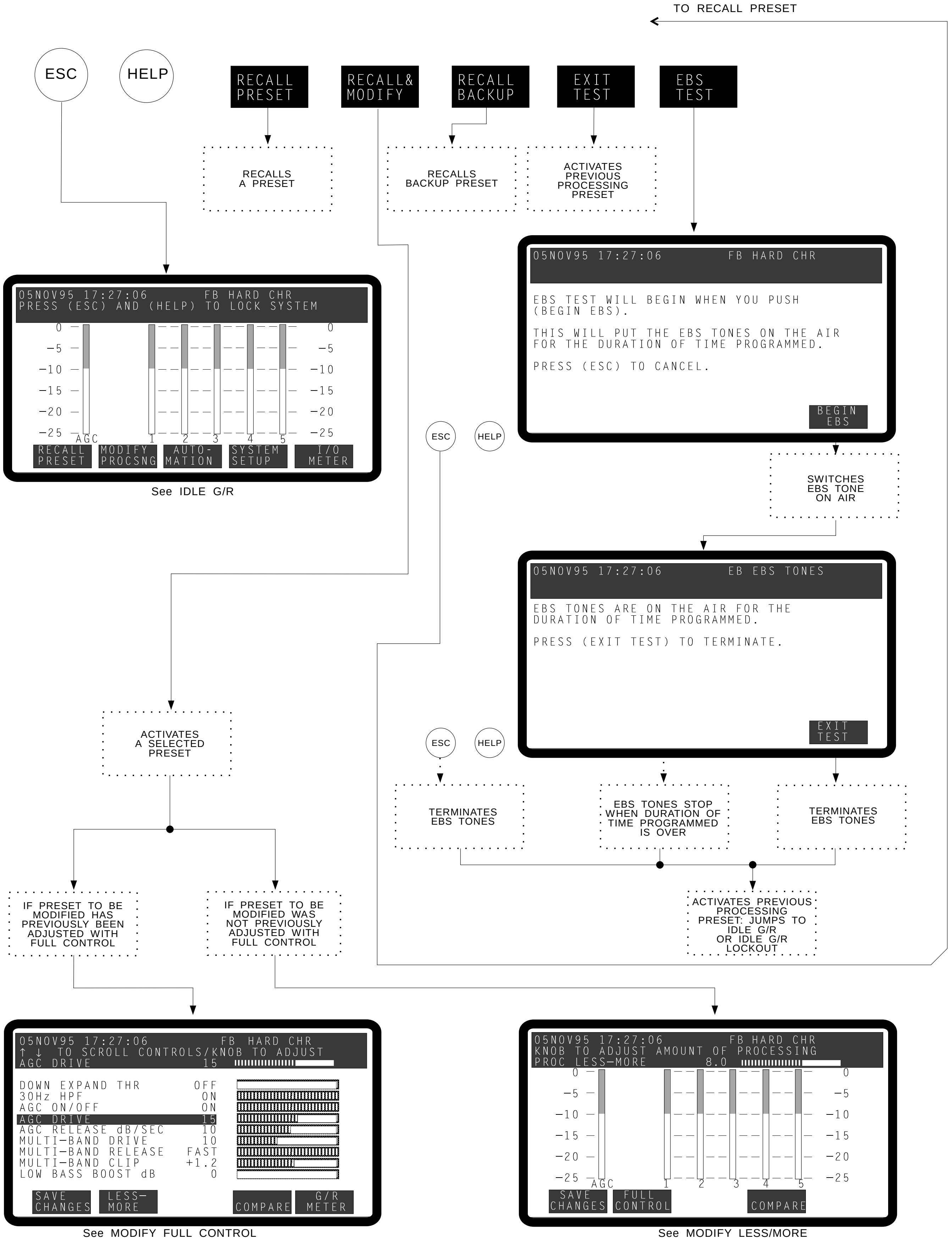
BEGIN EBS

Press to activate EBS TEST for duration programmed.

EXIT TEST

EXIT TEST is displayed on the screen only when the preset on the air is a test preset. Press to exit the test and activate the previous processing preset (not the previous test preset). (If the automation would have activated a new preset during the time you were using a test preset, EXIT TEST will activate this new preset, not the previous processing preset.)

RECALL PRESET Flow Chart



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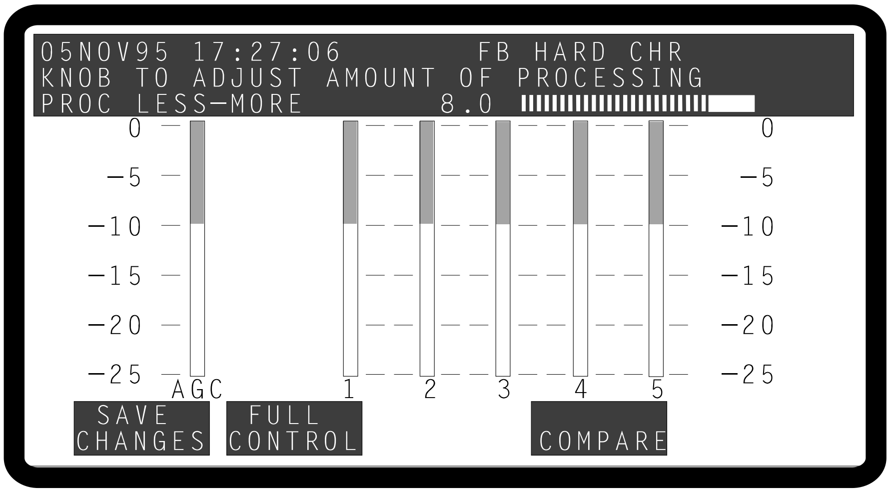
MODIFY Screens

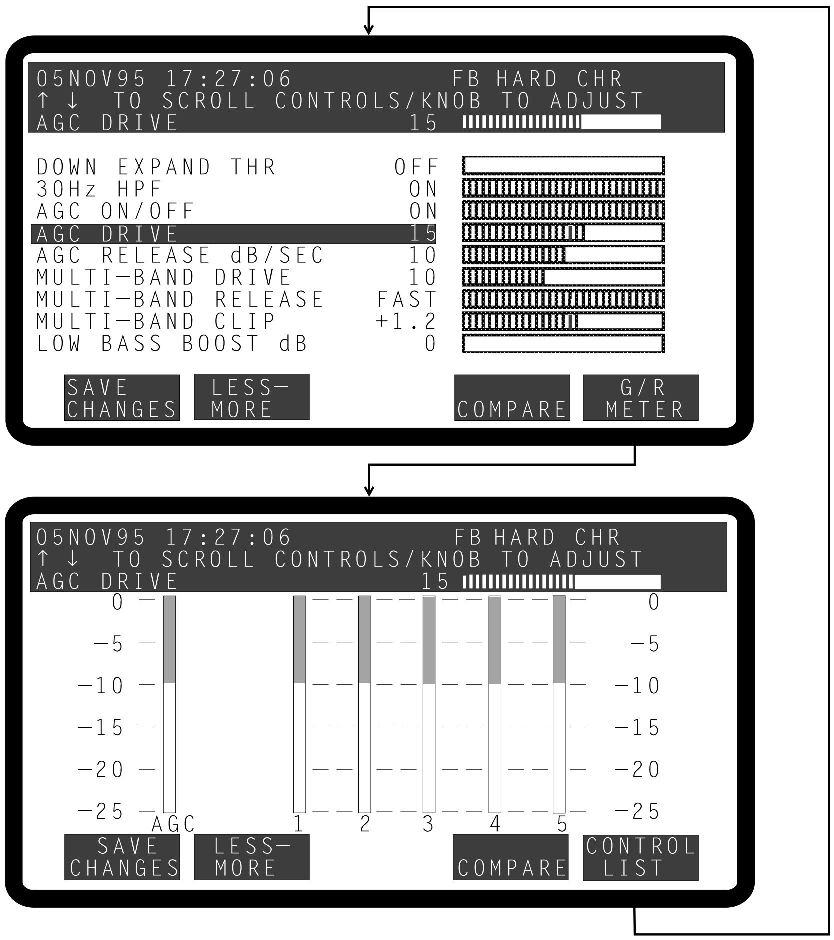
MODIFY LESS-MORE Screen



Modify Processing Less-More adjusts all of the controls precisely to factory recommended settings for various amounts of processing. It is a great way to adjust the sound on the air without having to understand what each of the many controls does.

Use the control knob to make less-more modifications. Note that as soon as the control knob is used, the ## PRESET NAME indication on line 1 changes to MODIF ## PRESET NAME to indicate a change to that preset. You can find your way back to the original setting by adjusting until the “MODIFY” flag (in front of the preset name) is extinguished.

MODIFY FULL CONTROL and MODIFY FULL G/R Screens



The MODIFY FULL screens let you adjust individual controls. The number and box next to each control show where the control is set.

The MODIFY FULL CONTROL screen lists each control and its setting, while the MODIFY FULL G/R screen shows gain reduction meters for the processing structure in use. Both screens show the currently-selected control and its setting on the third line of the screen display. This allows for easy adjustment from either screen.

Use the ↑ and ↓ keys to select a control from the list. Change the settings of that control with the front panel control knob. If any control is changed, an “M” is displayed to the right of the meter to indicate a change to that control and the ## PRESET NAME indication on line 1 changes to MODIF ## PRESET NAME to indicate a change to that preset. Controls can be set back to their original positions by observing the “M” flags while adjusting.

Press the fifth soft key (G/R METER or CONTROL LIST) to toggle between the two screens.

MODIFY FULL CONTROL , MODIFY FULL G/R and MODIFY LESS-MORE Buttons

SAVE CHANGES

When you adjust any of the processing controls, the sound on the air changes. But these changes are not yet stored in a preset. Pressing SAVE CHANGES lets you store the new settings in the same or a different preset.
See SAVE CHANGES (page 3-57).

LESS-MORE

From MODIFY FULL CONTROL screen, returns you to MODIFY LESS-MORE to change all of the controls together, to the factory-recommended settings.
But if you have changed any controls with FULL CONTROL, you cannot return to MODIFY LESS-MORE without losing your customized settings. You will first be warned that you will lose your FULL CONTROL control settings.

FULL CONTROL

FULL CONTROL is for the station that wants to fine-tune the sound beyond the factory recommended settings in LESS-MORE.
Once you change the sound with FULL CONTROL, you cannot return to MODIFY LESS-MORE without losing your customized settings.

COMPARE NEXT

COMPARE NEXT is used for switching between a selected (highlighted) preset and a modified preset.
The second line of the screen display shows what preset will be active if you press this button.
See COMPARE (page 3-56).

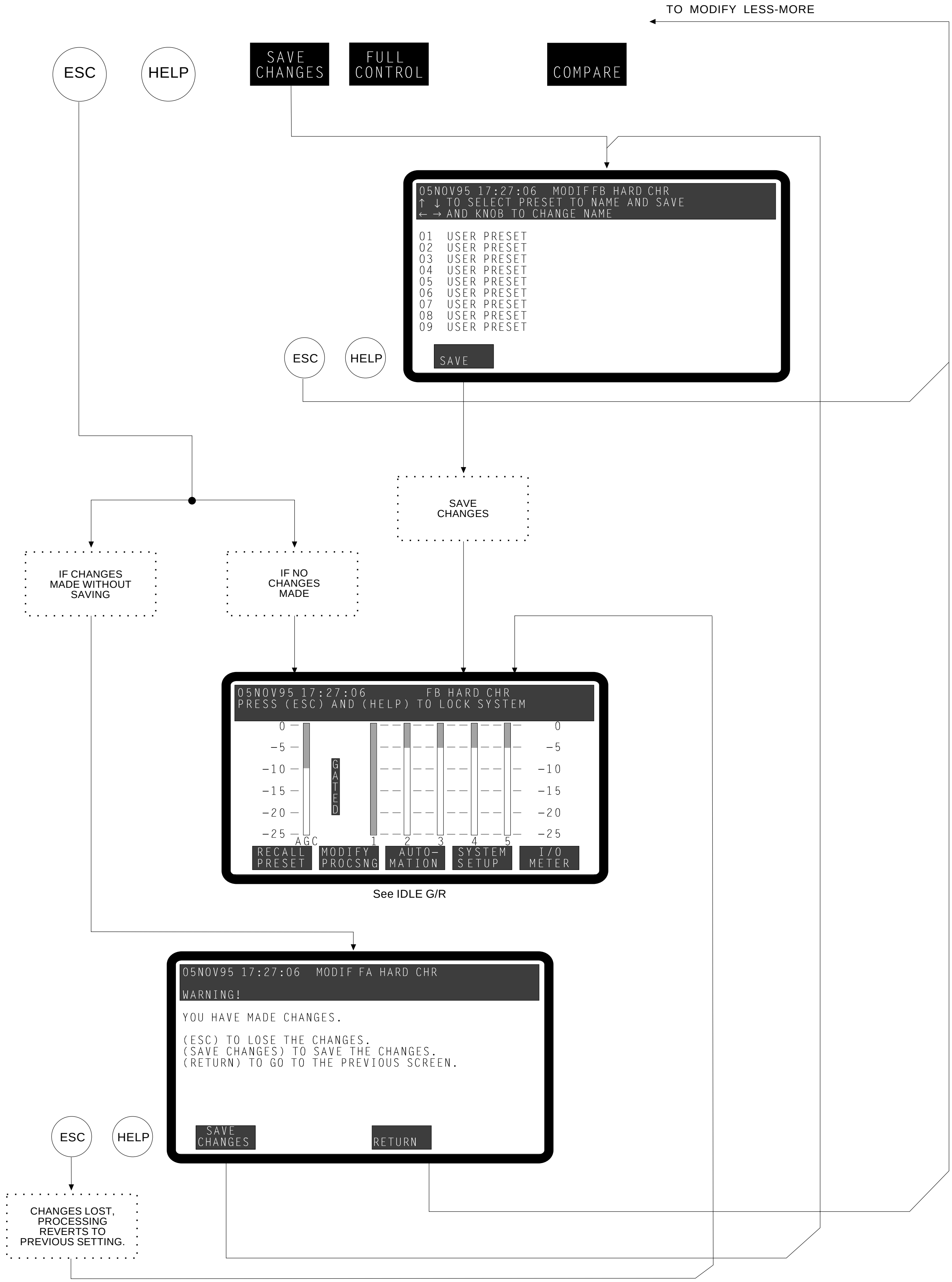
G/R METER

From MODIFY FULL CONTROL screen, press G/R METER to see gain reduction levels instead of full control list. Metering will show gain reduction for the processing structure in use.
You can still scroll the controls and change their settings. The selected control name and its setting are shown above the meters (on the third line of the screen display).
Press the far-right soft key (CONTROL LIST or G/R METER) to toggle between the two screens.

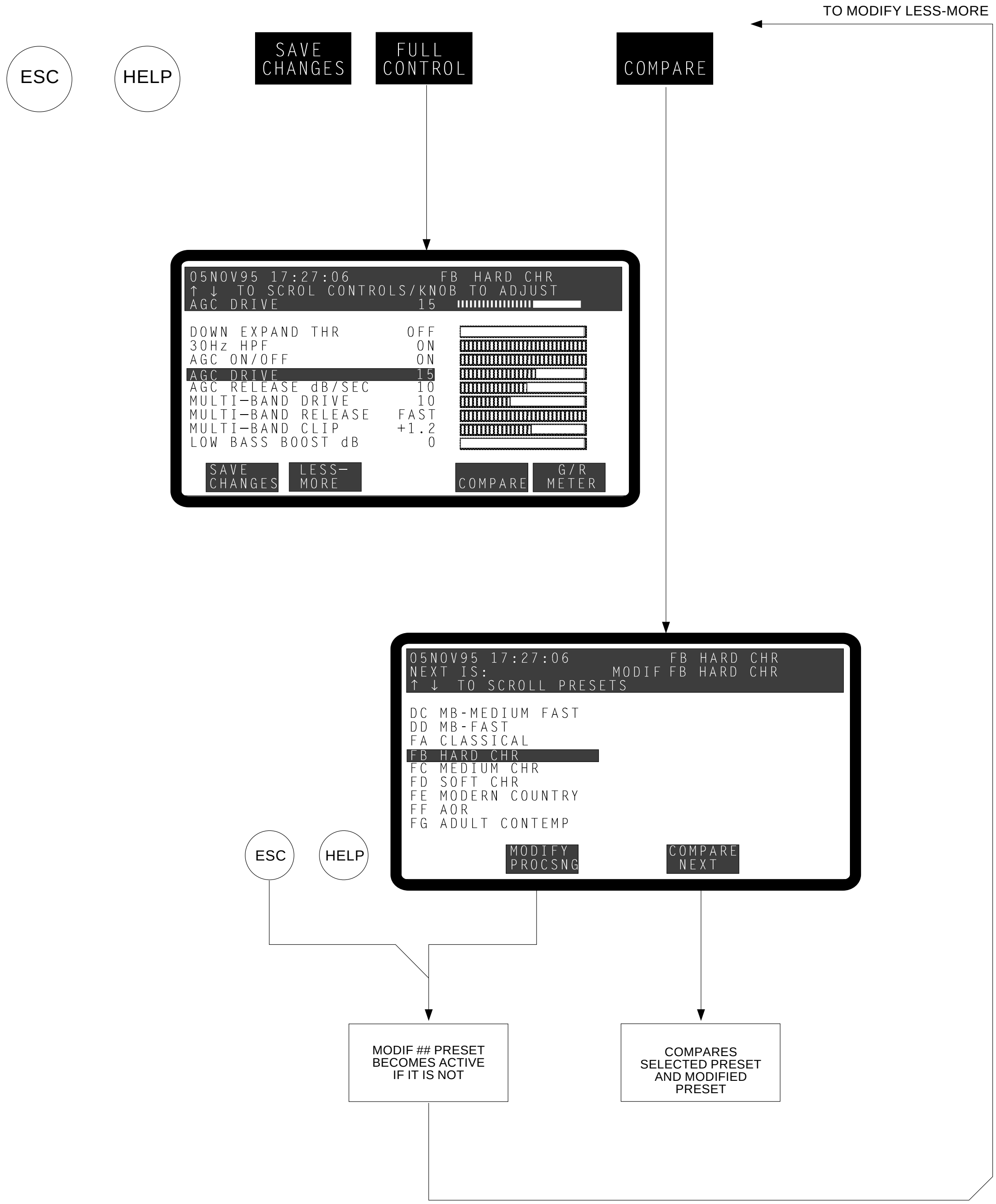
CONTROL LIST

From MODIFY FULL GAIN REDUCTION screen, press CONTROL LIST to see full list of controls instead of gain reduction meters.
The selected control name and its setting are highlighted in the list, and are also shown on the third line of the screen display.
Press the far-right soft key (CONTROL LIST or G/R METER) to toggle between the two screens.

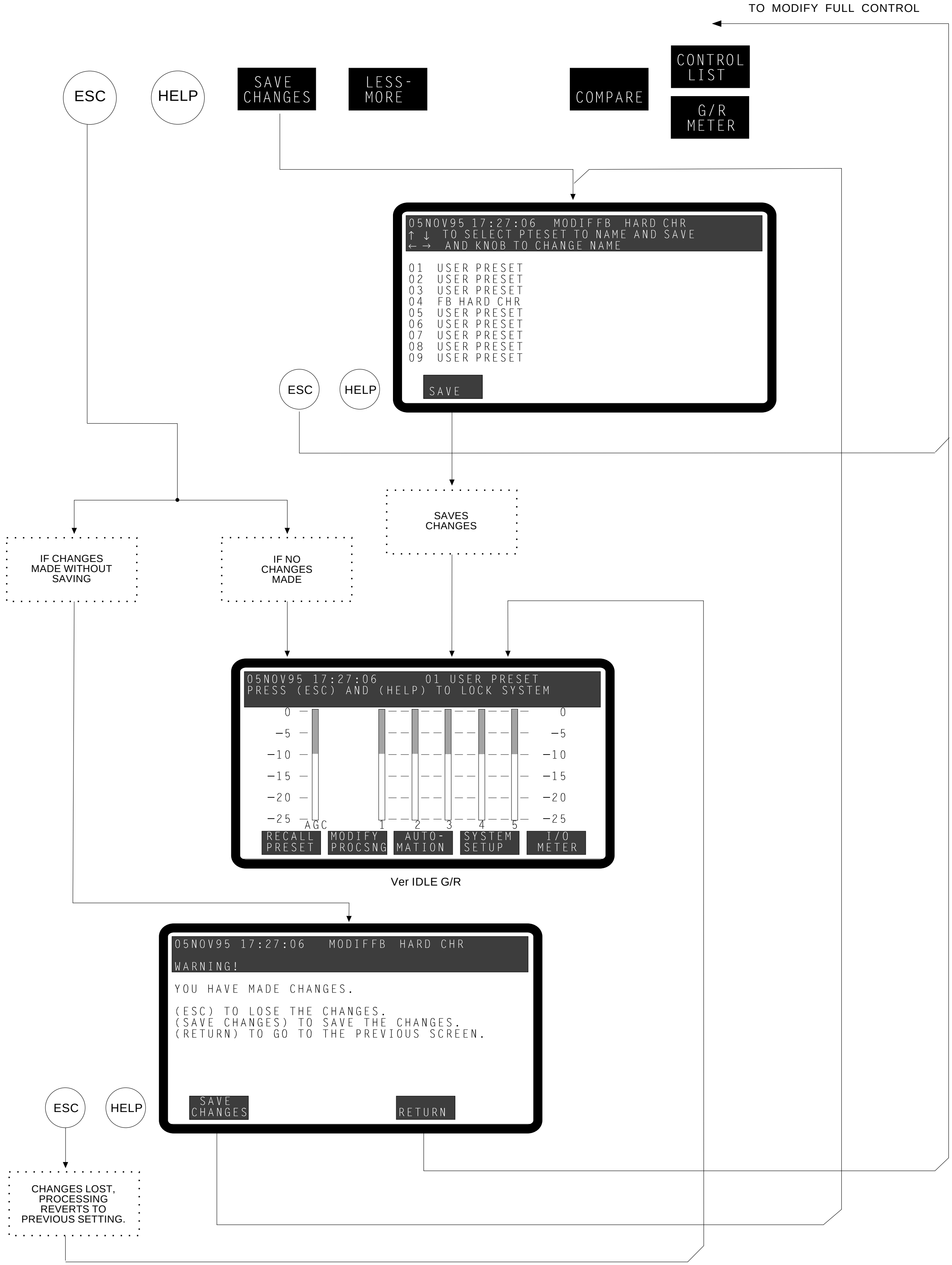
MODIFY LESS-MORE Flow Chart



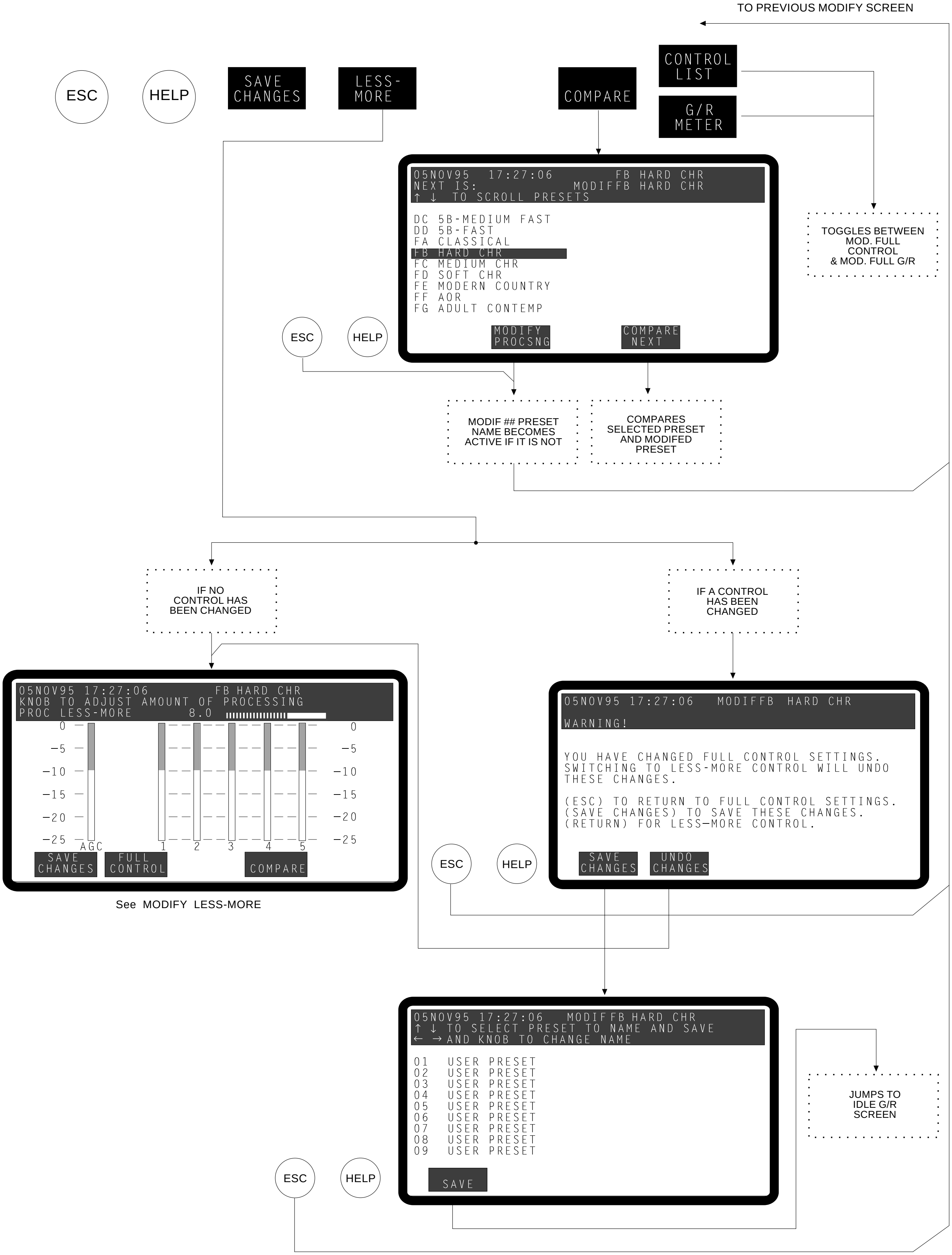
MODIFY LESS-MORE Flow Chart



MODIFY FULL CONTROL and MODIFY FULL G/R Flow Chart

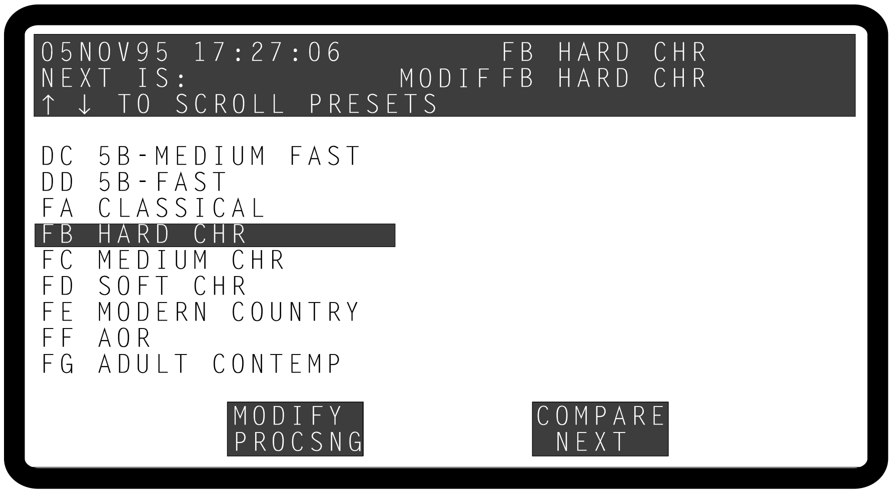


MODIFY FULL CONTROL and MODIFY FULL G/R Flow Chart



See MODIFY LESS-MORE

COMPARE Screen and Buttons



MODIFY PROCSNG — Returns you to the previous screen (`MODIFY LESS-MORE`).

COMPARE NEXT — COMPARE is used for switching between a selected (highlighted) preset and a modified preset.
Use the ↑ and ↓ keys to scroll to a preset.
The second line of the screen display shows what preset will be active if you press this button.

SAVE CHANGES Screen and Button

```
05NOV95 17:27:06   MODIFFB HARD CHR
↑ ↓ TO SELECT PRESET TO NAME AND SAVE
← →   AND KNOB TO CHANGE NAME

00   USER PRESET
01   USER PRESET
02   USER PRESET
03   USER PRESET
04   USER PRESET
05   USER PRESET
06   USER PRESET
07   USER PRESET
08   USER PRESET

    SAVE
```

SAVE

When you adjust any of the processing controls, the sound on the air changes. These changes are not stored in a preset until you press SAVE from this screen.

Use the ↑ and ↓ keys to select preset # to name and save.
Use ← and → keys to select each character, and the control knob to change the character. If you do not assign a new name, the name in the upper right corner of the screen is assigned.

Press SAVE to store the new settings in the selected preset.

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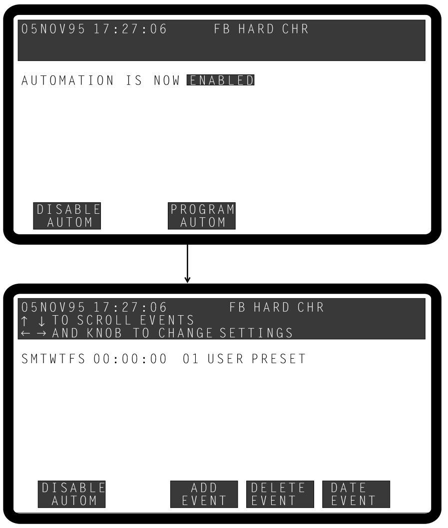
AUTOMATION Screens

AUTOMATION Screen and Buttons

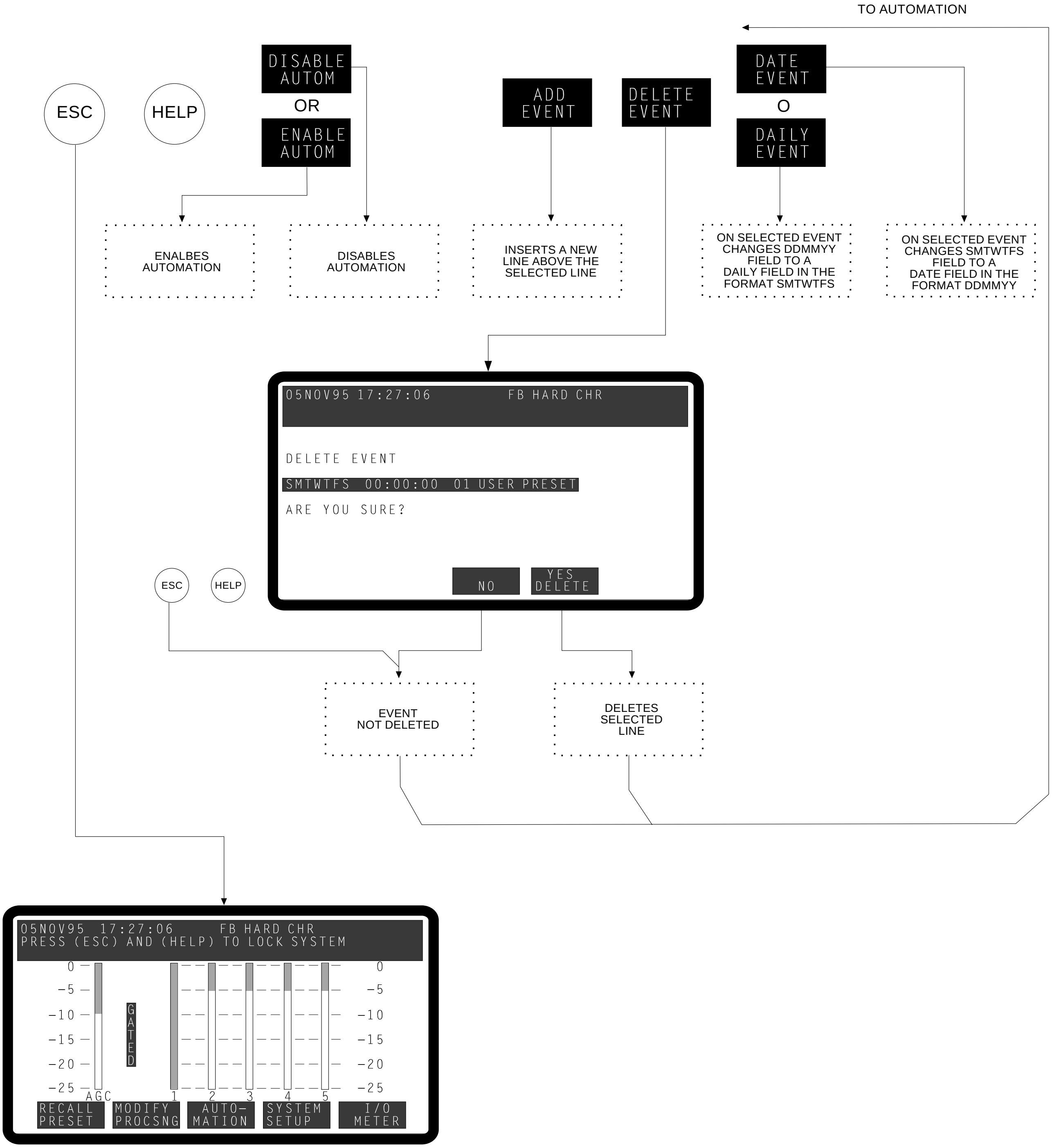


Automation is automatic programmed change of presets at scheduled times. The `AUTO PROGRAM` screen shows a listing of all automation events programmed.
Use the ↑ and ↓ keys to scroll up and down event list.
Use ← and → keys to move through automation field, and the control knob to make changes.

You can set the event to happen by day(s) of the week, or on one specific date.

DISABLE AUTOM	Disables Automation. This button appears if Automation is enabled.
ENABLE AUTOM	Enables Automation. This button appears if Automation is disabled.
PROGRAM AUTOM	Access `AUTO PROGRAM` screen. This button is displayed if pascode permits access to program `AUTO PROGRAM` screen.
ADD EVENT	Inserts a new line above the selected line. Use ← and → and the control knob to program the event.
DELETE EVENT	Deletes an event from Automation. You are first given a warning, however.
DATE EVENT	On selected event, changes SMTWTFS field to a date field in the format DDMMYY. This button appears if the highlighted event is a daily field.
DAILY EVENT	On selected event, changes DDMMYY field to a daily field in the format SMTWTFS. This button appears if the highlighted event is a date field.

AUTOMATION Flow Chart



See IDLE G/R

AUTOMATION Additional Screen Information

Using DATE EVENT and DAILY EVENT to Automatically Change Presets.

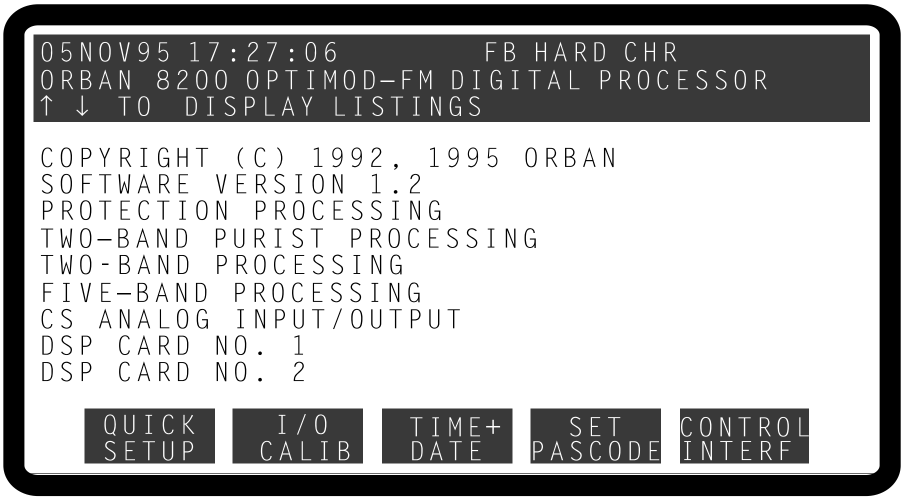
The 8200's automation software will allow a DATE EVENT to take precedent over a DAILY EVENT. This means that event changes automated by DAILY EVENT are activated as scheduled until the event specified with DATE EVENT, then the DATE EVENT overrides any previously scheduled DAILY EVENT programmed for the same time. After the DATE EVENT takes place, the next DAILY EVENT will resume its schedule. This eliminates the need for the user to delete the scheduled DAILY EVENT when DATE EVENT is programmed.

WARNING: Unless you are intentionally overriding a DAILY EVENT with a DATE EVENT, do not program two events to occur at the same time of the day. The software only recognizes the first setting listed in the event list. The second setting listed will be ignored.

IMPORTANT: If you want to activate a processing preset or test preset and an output mode (STEREO, MONOL, MONOR) **simultaneously**, insure that the time settings in the EVENT list are at least 1 second apart.

SYSTEM SETUP Screens

SYSTEM SETUP Screen and Buttons



The screen shows: Software version number and serial number
Which processing structures the software contains
Hardware that is installed.

QUICK SETUP

QUICK SETUP gets you on the air fast by walking you through initial setup procedure, step by step. To make system changes after initial setup, use I/O CALIB and other system controls, as necessary.

I/O CALIB

Press I/O CALIB to access a detailed listing of all system calibrations. Most of these are set in the QUICK SETUP procedure.

TIME+ DATE

Press TIME+ DATE to access controls to set the time, date, and dates for daylight savings time, if used in your country.
Use the ↑ and ↓ keys to scroll preset list up and down. Use the ← and → keys on TIME or DATE settings to move through hour-minute-second or month-day-year. Change parameters with the control knob. Press ENTER CHANGE to store these changes in the system. Or press ESC to ignore the changes.

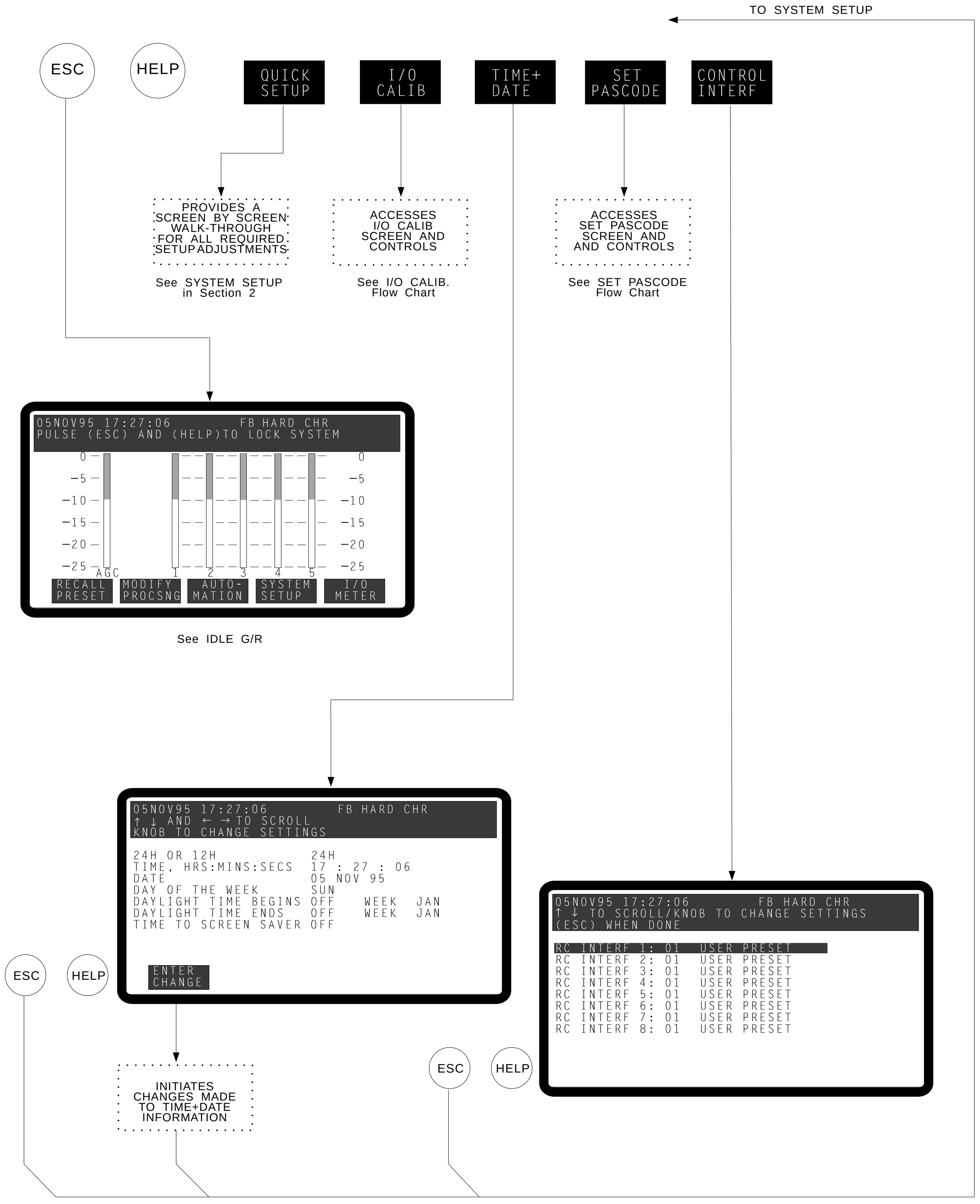
SET PASCODE

Press SET PASCODE to access pascode controls to program the security lockout of the system, and program the pascodes.

CONTROL INTERF

Press CONTROL INTERF to program the remote control interface and its eight (8) opto-isolated inputs. Each can be programmed to recall any preset, or switch to Stereo, Mono Left, Mono Right, reduce Main/Stereo level when SCA1 or SCA2 are ON, reduce Main/Stereo level when TX1 or TX2 are ON. Note: remote functions that reduce Main/Stereo level do so only when current is applied to the appropriate input, so these functions must be externally held active — they do not latch.

SYSTEM SETUP Flow Chart (QUICK SETUP, TIME+ DATE, INTERFACE)



I/O CALIB and I/O CALIB I/O METER Screen Buttons

Press far-right key (`I/O METER` or `CONTROL LIST`) to toggle between the two screens.

MODIFY EBS — Access `MODIFY EBS` screen to modify EBS Test Tone. Accessing this screen does not implement EBS test (see page 46). Press ESC to store changes.

PILOT LEVEL — Pilot Level is shown on a meter from 7.5% to 10.5%.
Adjust pilot level on the stereo encoder with the screwdriver slot in the front panel. See pilot level metering information on page 4-14.

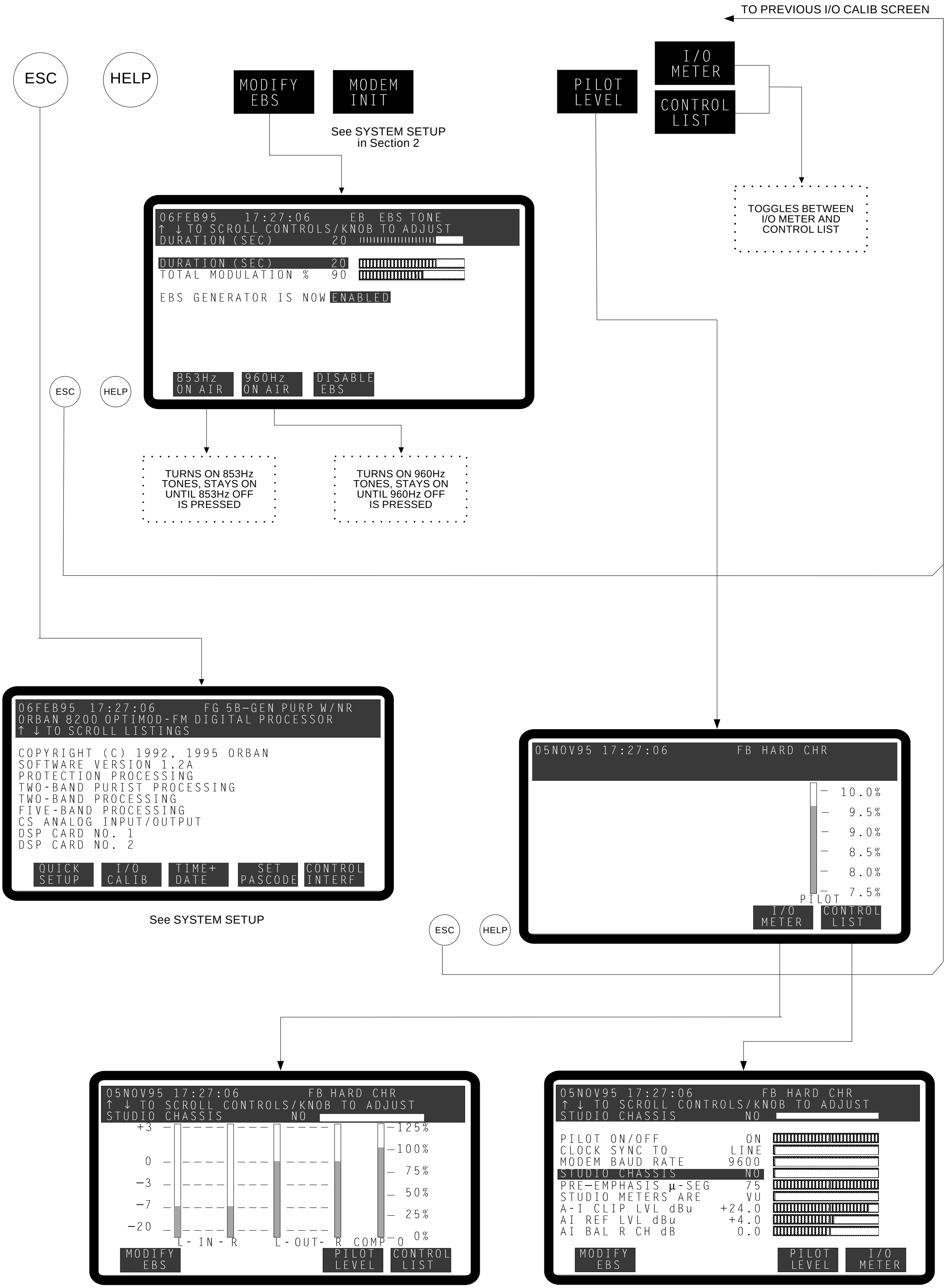
I/O METER — From `I/O CALIB` (or `PILOT LEVEL`) screen, changes the screen to show Input/Output levels.
The selected control is shown above the meter.
Use the ↑ and ↓ keys to scroll input/output list. Adjust the level or parameter with the control knob.

CONTROL LIST — From `I/O CALIB` or `PILOT LEVEL` screen, changes the screen to show the full list of controls instead of the Input/Output meters.
Use the ↑ and ↓ keys to scroll input/output list. Adjust the level or parameter with the control knob.

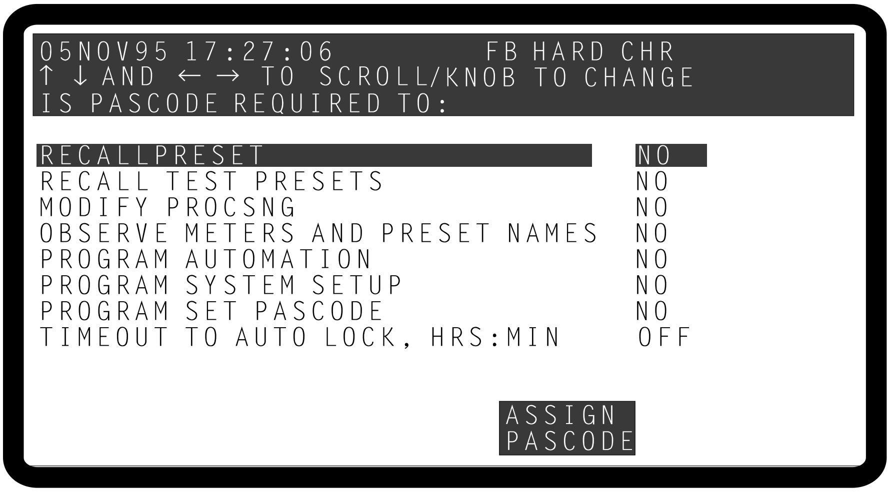
853HZ ON AIR — From `MODIFY EBS` screen, enables 853Hz tone. This tone stays on air until 853Hz OFF button is pressed.

960HZ ON AIR — From `MODIFY EBS` screen, enables 960Hz tone. This tone stays on air until 960Hz OFF button is pressed.

I/O CALIB CONTROL and I/O CALIB I/O Flow Chart



SET PASCODE Screen



The `SET PASCODE` screen provides access to the controls and screens necessary to program the security lockout of the system and program the pascodes.

Use this screen to select functions for lockout.
Use the `ASSIGN PASCODE` screen to assign functions to a particular pascode number.

Setting the security status of the 8200

All of the settings on this screen determine whether pascodes are required to access the various functions listed. If a function is listed as NO, the function can be accessed and changed from the front panel without the user having to enter a pascode. If a function is listed as YES, the user must first enter a pascode, and that pascode must have been previously programmed to permit access to that function.

When a pascode is entered, only those functions that are permitted by that pascode are displayed on the screen.

Use the ↑ and ↓ arrow keys to scroll through and select (highlight) the settings. Use the front panel control knob to change the setting.

ASSIGN PASCODE

When you are finished setting the security status of the 8200, press the ASSIGN PASCODE button to access the `ASSIGN PASCODE` screen.

If you do not wish to program pascodes at this time, press the ESC button to return to the `SYSTEM SETUP` *screen. Press the ESC button again to return to the* `IDLE G/R` *screen.*

SET PASCODE Screen Parameters

- RECALL PRESET

  [NO] or [YES]

  Selects whether a pascode (authorized to RECALL PRESET) is required to recall processing presets.

- RECALL TEST PRESETS

  [NO] or [YES]

  Selects whether a pascode (authorized to RECALL TEST TONES) is required to recall test presets, such as TONE and BYPASS.

  EBS (Emergency Broadcasting System) can be accessed at all times, with or without a pascode.

  NOTE: If the security status is set so that a pascode is required to recall test presets, and if a test preset has been stored as a user preset, it will be locked out to users not authorized to recall test presets.

- MODIFY PROCSNG

  [NO] or [YES]

  Selects whether a pascode (authorized to MODIFY PROCSNG) is required to modify the processing (change the sound of the station) and save those changes to a user preset.

- OBSERVE METERS AND PRESET NAME

  [NO] or [YES]

  Selects whether a pascode (authorized to observe meters and preset name) is required to display the metering on the screen. Some stations may wish to restrict meter display from the screen so that their “setup” cannot be copied by another station.

- PROGRAM AUTOMATION

  [NO] or [YES]

  Selects whether a pascode (authorized to PROGRAM AUTOMATION) is required to change the automation event schedule.

- PROGRAM SYSTEM SETUP

  [NO] or [YES]

  Selects whether a pascode (authorized to change SYSTEM SETUP) is required to access QUICK SETUP, I/O CAL, TIME+ DATE and CONTROL INTERF.

● PROGRAM SET PASCODE

[NO] or [YES]

Selects whether a pascode (authorized to program SET PASCODE) is required to access SET PASCODE.

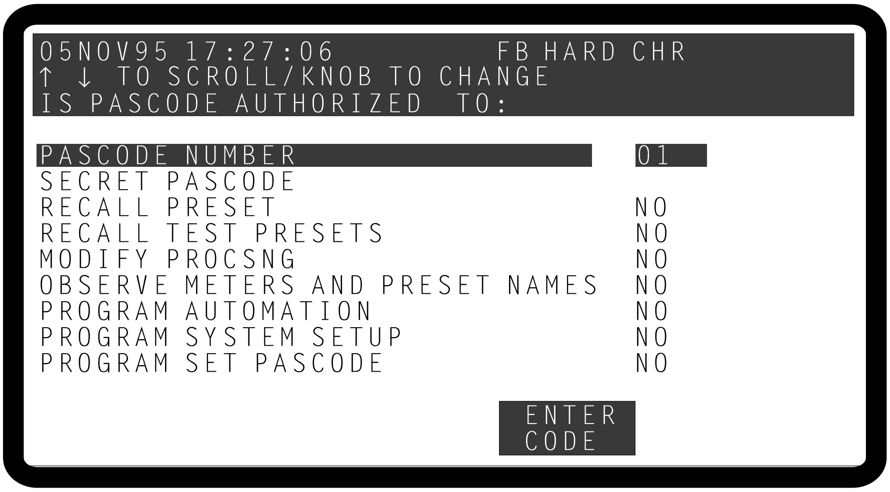
NOTE: A pascode authorized for System Setup will automatically be given access to all other functions, as seen on the screen.

● TIMEOUT TO AUTO LOCK, HRS:MINS

[0:10] to [8:00] or [OFF]

Timeout automatically locks the system after the period of time set by this control. We strongly recommend that you set timeout to a reasonable time, so that if the 8200 if left unattended, it will lock itself.

ASSIGN PASCODE Screen



The ASSIGN PASCODE screen provides the controls necessary to program the pascodes.

Use the ASSIGN PASCODE screen to assign (or "authorize") functions to a particular pascode number (after using the SET PASCODE screen to select functions for lockout).

<u>Assigning functions to a particular pascode number:</u>

The 8200 allows you to program up to 10 pascodes. Each pascode can be programmed to allow or deny access to the various functions listed.

PASCODE NUMBER is used to select which of the 10 pascodes you will program.

The rest of the lines show which functions that pascode is allowed to access. Note that if you previously programmed the SET PASCODE screen parameters so that a pascode is not required for a given function, the user will be allowed access to that function, even if the pascode is programmed NO for that function from the ASSIGN PASCODE screen.

Use the ↑ and ↓ arrow keys to scroll through and select (highlight) the settings. Use the front panel control knob to change the setting.

ENTER CODE

Press to access ENTER CODE screen, then enter the desired pascode, up to eight digits. From the ENTER CODE screen, press ESC to store the entered pascode and return to the previous screen.

Note that if a pascode has been previously entered, and you simply escape the ENTER CODE screen, the previously entered pascode will remain valid.

ASSIGN PASCODE Screen Parameters

- PASCODE NUMBER

  [1] to [10]

  Selects which of ten pascodes to program authorization and code. As this number is changed, the other fields will change to reflect the authorization of that code.

  SET PASCODE allows you to program up to 10 secret pascodes, each with one to eight digits.

- SECRET PASCODE

  [1] to [99999999]

  Allows you to set pascode with one to eight digits.

- RECALL PRESET

  [NO] or [YES]

  Selects whether this pascode (PASCODE NUMBER displayed on the screen) is authorized to recall processing presets.

- RECALL TEST PRESETS

  [NO] or [YES]

  Selects whether this pascode (PASCODE NUMBER displayed on the screen) is authorized to recall test presets, such as TONE and BYPASS.

  EBS (Emergency Broadcasting System) can be accessed at all times, with or without a pascode.

  NOTE: If the security status is set so that a pascode is required to recall test presets, and if a test preset has been stored as a user preset, it will be locked out to users not authorized to recall test presets.

- MODIFY PROCSNG

  [NO] or [YES]

  Selects whether this pascode (PASCODE NUMBER displayed on the screen) is authorized to modify the processing (change the sound of the station) and save those changes to a user preset.

- OBSERVE METERS AND PRESET NAME

  [NO] or [YES]

  Selects whether this pascode (PASCODE NUMBER displayed on the screen) is authorized to observe meters and preset name.

- PROGRAM AUTOMATION (implemented in software version 1.0 and higher)

  [NO] or [YES]

  Selects whether this pascode (PASCODE NUMBER displayed on the screen) is authorized to change the automation event schedule.

- PROGRAM SYSTEM SETUP

  [NO] or [YES]

  Selects whether a pascode (authorized to change SYSTEM SETUP) is required to access QUICK SETUP, I/O CAL, TIME+ DATE and CONTROL INTERF.

  NOTE: A pascode authorized for System Setup will automatically be given access to all other functions, even if that pascode is not programmed for those other functions.

- PROGRAM SET PASCODE

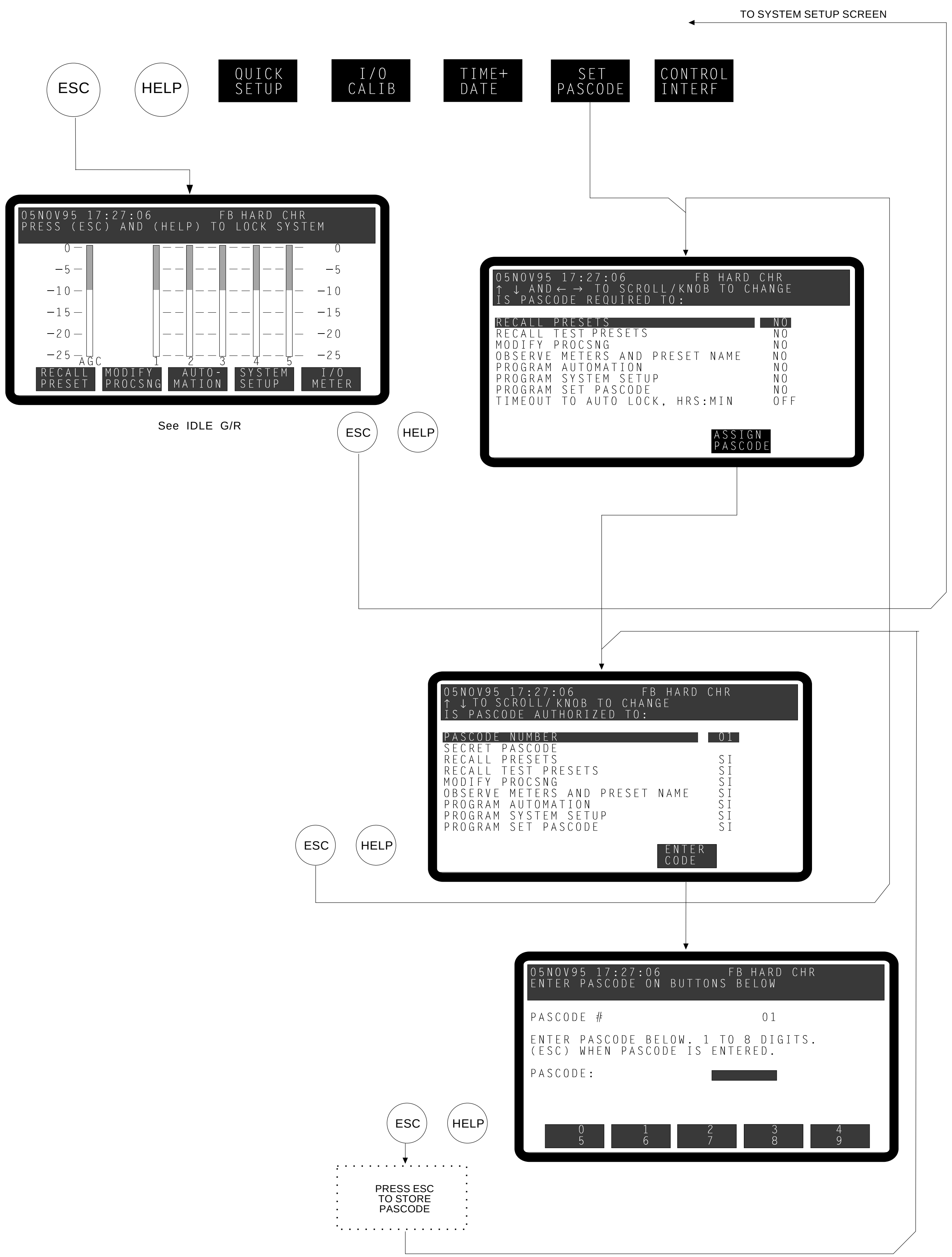
  [NO] or [YES]

  Selects whether a pascode (authorized to program SET PASCODE) is required to access SET PASCODE.

  NOTE: A pascode authorized for SET PASCODE will automatically be given access to all other functions, even if that pascode is not programmed for those other functions.

  NOTE: At least one of the ten pascodes must be authorized to program set pascode. This user has ultimate control of the 8200's security, and the unit will not lock until this requirement is fulfilled.

SET PASCODE Flow Chart



LOCKOUT Screen

The ENTER PASCODE button signifies that at least one 8200 front panel control function is locked (i.e., requires a pascode for access, according to the access permission set in SET PASCODE and the authorization for the pascode entered). If all functions are locked, the screen appears as shown above. If a function is not locked, it will be shown. RECALL PRESET button appears even when RECALL PRESET is locked. (But when RECALL PRESET is pressed, if RECALL PRESET is locked, only the EBS TEST button appears on the screen). MODIFY PROCSNG appears when MODIFY PROCSNG is allowed by pascode. AUTOMATION appears if either AUTOMATION or RECALL PRESET are allowed by pascode (But when AUTOMATION is pressed, if AUTOMATION is locked, only the DISABLE AUTOM/ENABLE AUTOM button appears on the screen). In LOCK-OUT mode, SYSTEM SETUP is always locked. Gain Reduction and Input/Output meters appear when OBSERVE METERS is allowed by pascode.

To access locked functions, you need to enter a valid pascode. If a pascode with limited access has been entered, buttons for functions not permitted access by that pascode will not be displayed.

LOCKOUT Screen Buttons

RECALL PRESET

Presets are stored settings of the processing controls (See Section 1). Press RECALL PRESET to see a list of all stored presets. You can select a preset from the list, and recall that preset on the air. If RECALL PRESET is locked, the list will be empty, but the EBS TEST button will be available to access the EBS Test Tone.

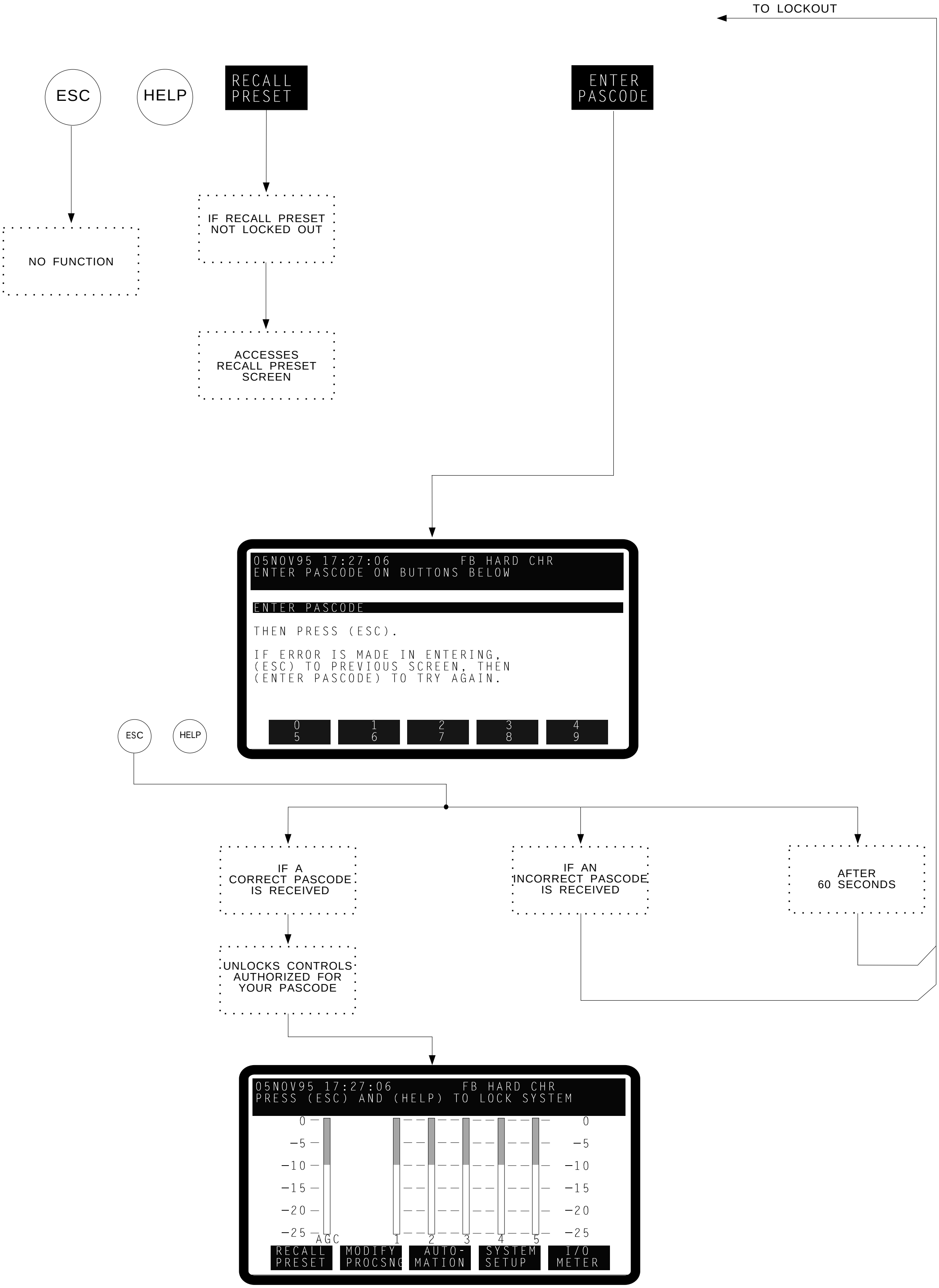
ENTER PASCODE

To enter your pascode:
1 - Press ENTER PASCODE.
2 - Enter your pascode on the five buttons below the screen.
When a correct pascode is entered, the system will unlock and access the `IDLE G/R` screen (or a `LOCKOUT` screen with limited access). If you make a mistake, press ESC, then ENTER PASCODE to try again. If you do not wish to enter a pascode, simply press ESC to return to the `LOCKOUT` screen.

EXIT TEST

From `LOCKOUT` screen, press EXIT TEST to exit the test preset on air and recall previous processing preset. This button is displayed if the preset on air is a test preset.

LOCKOUT Flow Chart



See IDLE G/R

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