AMB-5 Ambient Sensing Mixer/Multiplexer OPERATION GUIDE

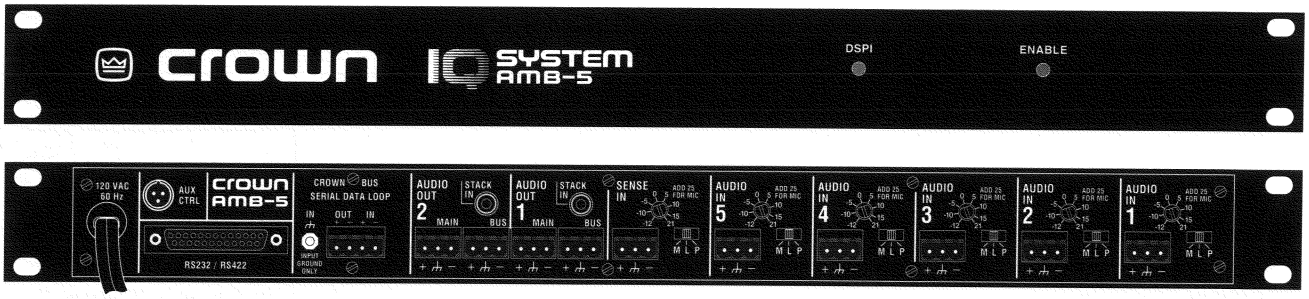
IQ-MSD

8/93

1 The *AMB-5*

The *AMB-5* is two mixer/multiplexers in one unit. One mixer/multiplexer is an ambient sensing 5x1 mixer/multiplexer. It has all the capabilities of Channel 1 of an *SMX-6* plus the added feature of ambient sensing. Its custom AMB *algo* enables it to automatically adjust the level of its mix to match the ambient noise level. The second mixer/multiplexer is a basic 6x1 mixer/multiplexer much like Channel 2 of an *MPX-6*. It offers remote control of Channel 2 from the host computer but it has none of the automatic features like those provided by the *algos* of Channel 1.

Like the *SMX-6* and *MPX-6*, the *AMB-5* has six mic/line inputs, two main line-level outputs and two switchable bus outputs. It contains a sensing circuit at the beginning of each input to sense the input signal level ahead of any signal processing. Similar sensors are located at each output channel. The sixth input is dedicated as a sensing input for connection to an ambient microphone and is used to input the ambient sound level for the AMB *algo* of Channel 1. It is also possible to use the sixth input as a "dumb" *MPX-6* input for Channel 2.



The on-board intelligence of the *AMB-5* enables it to perform many versatile functions like automatic mixing, compression, limiting, automatic level control and ambient-tracking level control. However, these features are available for Channel 1 only. Channel 2 functions like an *MPX-6* with manual switching and routing capabilities.

The *AMB-5* is ideal for any installation where background noise levels vary throughout the day. In factories, its ambient sensing capability enables it to adjust its output level to match the noise level of the factory floor from one shift to the next. In a train station it can automatically sense that a loud train is arriving, turn up the level appropriately so announcements can be heard and then turn the level back down when the train has departed. It can also safeguard against feedback and other common audio problems.

Because the *AMB-5* is configured and controlled by computer software, its function is limited only by the sophistication and imagination of the programmer. Buy you don't have to be a computer hack to unlock the power of a sensing multiplexer. The straight-forward settings in the individual *algos* which comprise the *AMB-5* algo pack and the logical organization of the IQ software *dataframe* make it very easy for a non-programmer to set up and operate an *AMB-5*.

1.1 The *AMB-5* Algo Pack

The on-board intelligence of the *AMB-5* is controlled by the algo pack in its microprocessor. The algo pack contains the individual *algos* which control its various automation features. Each *algo* is listed below along with a brief description.

- ❑ **Gate *algo*:** Automatically turns an input on and off (called "gating") whenever an incoming audio signal level passes a preset threshold level.
- ❑ **Duck Priority *algo*:** Controls the priority of each input and how much the low priority inputs will be attenuated or "ducked" when a higher priority input is activated (its gate is opened). There are six different priority levels.
- ❑ **Input Compressor/Limiter *algo*:** Controls the threshold, compression ratio and release time of a feed-forward input compressor/limiter. There is a separate Channel 1 and 2 compressor/limiter for each input.
- ❑ **Auto Level *algo*:** Automatically adjusts the level of selected inputs to achieve a target level. Each input has a separate Channel 1 and 2 target level.
- ❑ **Output Compressor/Limiter *algo*:** Controls the threshold, compression ratio and release time of each channel's output compressor/limiter.
- ❑ **Gate Count *algo*:** Counts how many inputs are activated at any instant and applies a preset attenuation as the number increases to prevent feedback.
- ❑ **AMB *algo*:** Monitors the level of the sense input and automatically adjusts the output level of Channel 1 up or down as appropriate.

All *algos*, except the AMB *algo*, can be individually turned on and off for each input, making them extremely versatile. This is controlled from the IQ software sub-block screens for each unit. As a group, all *algos* in the *AMB-5* algo pack can be turned on or off with the ASA control on the *AMB-5* control block screen.

The *AMB-5* contains a built-in battery which preserves all algo pack settings for at least 60 days (when fully charged). This enables you to configure an *AMB-5* before it is physically installed into an audio system. *Note: It is a good idea to plug an AMB-5 into an active AC receptical for 24 hours before programming it to make sure its battery is fully charged.*

What is a feed-forward compressor?

Most compressors are "feedback" compressors. This means they watch the output level and if it rises above a preset threshold the compressor adds input attenuation. In this way it feeds back from the output to the input. The Output Compressor/Limiter *algo* works like this.

On the other hand, a "feed-forward" compressor watches the input and if it rises above a preset threshold the compressor adds attenuation downstream. In this way it feeds forward along the audio path. The Input Compressor/Limiter *algo* operates like this.

1.2 *AMB-5* Features

Manual controls for Channel 1 & 2 (software):
The manual software features are controlled on the control block screen.

- ❑ Increase and decrease each input level in ½ dB steps over a 125 dB range.
- ❑ Independently switch each audio bus output on and off.
- ❑ Independently mute each input (mute = full attenuation).
- ❑ Turn any auxiliary device connected to the AUX port on or off.
- ❑ Force the DSPI (Data Signal Presence Indicator) to remain on. (A trouble-shooting feature for Crown Bus loop communication.)

Manual controls for Channel 1 & 2 (hardware):
The manual hardware controls are located on the back panel of the unit.

- ❑ Individual three-position input level control for each input. The positions are for line, mic and phantom powered mic level signals.

- ❑ Individual preamp gain control for each input. (Can be used to compensate for differing input levels.)

Automation controls for Channel 1 only (software):
The automation software features are controlled in the sub-block screens.

Notes: The signal processing of each input can be controlled separately, providing excellent flexibility. The switchable audio bus output (Channel 1 only) receives the same post-processed signal that the main audio output receives.

- ❑ Gate logic and gate counting controls supervise automatic input switching and can prevent unwanted feedback.
- ❑ Input ducking controls supervise priority input mixing. ("Ducking" is the attenuation of an input to a preset "duck" level. This happens when an input with a higher priority is switched on by a gate *algo*.)
- ❑ Auto level controls work to achieve a target level for each input. Each input has a separate target level.
- ❑ Feed-forward input compressor/limiter and output compressor/limiter controls. Each input has a separate compressor/limiter.
- ❑ Ambient level sensing controls work to achieve a desired loudness above the ambient noise level.

Protection features:
- ❑ Automatic microprocessor reset if noise spikes, power dropouts or other disturbances are encountered.
- ❑ DSPI (Data Signal Presence Indicator) provides swift isolation of wiring problems, improper IQ addresses and disconnected or off-line IQ components.
- ❑ NiCad battery and built-in charger maintain all settings. When fully charged, the unit can be disconnected from power for 60+ days.

Digitally controlled analog circuitry:
- ❑ Because the analog circuitry is digitally controlled, each input can be processed by more than one *algo* at the same time without the unexpected side affects which would occur under analog control.
- ❑ Digital control also permits precise calibration. The gain settings of the *AMB-5* are accurate to better than ±1.0 dB.

Optional features:
- ❑ Option 1: The *AMB-5* can function as an IQ interface for one Crown Bus loop.

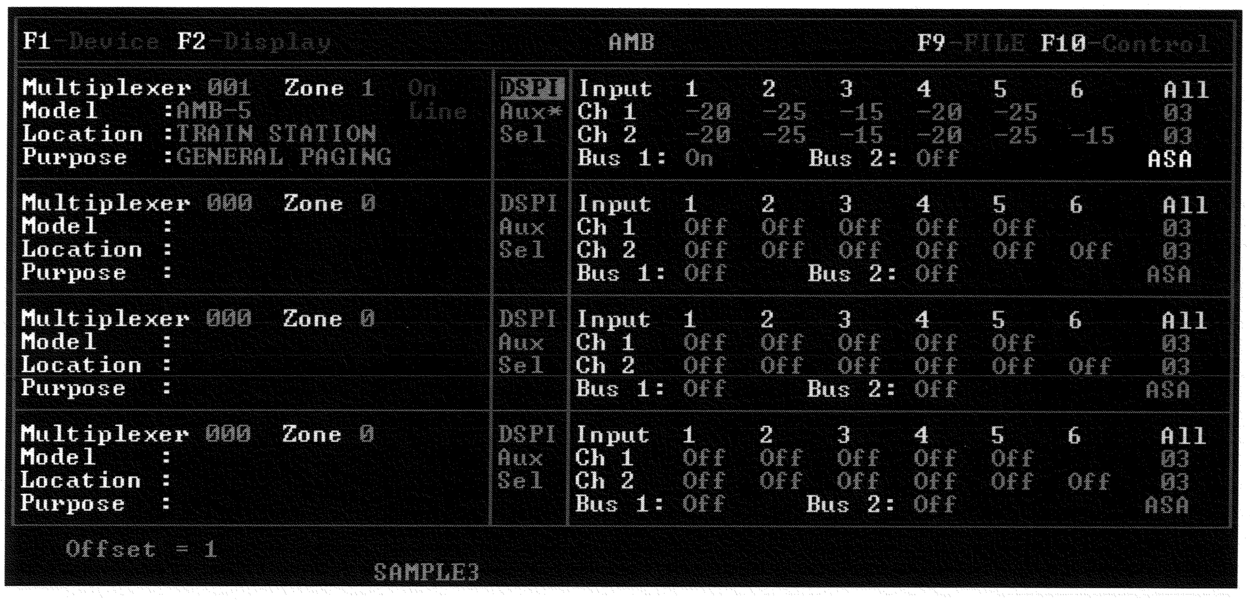
1.3 About this Operation Guide...

This *Operation Guide* describes how to use IQ software to configure and control an *AMB-5* mixer/multiplexer. It begins with manual operation in Section 2, describes the Display screen in Section 3 and delves into the configuration of its automatic features in Section 4. Afterward, several sample applications are presented in Section 5.

This *Operation Guide* does not discuss the hardware installation or hardware controls. Please refer to the *IQ Mixer/Multiplexer Hardware Installation Manual* for those details. Nor does this *Operation Guide* discuss the general features of the IQ software. Please refer to the IQ software *User's Manual* for that information.

2 Manual Control

Like an *MPX-6*, the *AMB-5* can be manually controlled with the control block screen of the IQ software. To switch to the *AMB-5* control block screen, press [F1] to access the Device menu then press [B]. [↑] [↓] can also be used to select the *AMB-5* control block screen from the Device menu followed by [Enter↵] to execute the selection. A mouse (🖱) can also be used. The *AMB-5* control block screen is shown below:



Please refer to the IQ software *User's Manual* for general instruction on how to navigate in the control block screen and how to select and change a control.

Tip: Because Channel 1 has two levels of control, manual and automatic, it is possible to have two separate preset mixes. This can be a great advantage when it is necessary to quickly switch to a second mix for Channel 1. One mix is created with the manual controls on the control block screen. The second mix is created with the *algo* controls on the sub-block screens. Toggling the ASA control on and off causes the *AMB-5* to switch between the two mixes. To toggle ASA on from the control block, select the ASA control and press [Spacebar]. To toggle ASA on and off from the sub-blocks, press [Ctrl]+[A]. *Note: The ASA control has no affect on Channel 2.*

A description of each control in an *AMB-5* control block follows:

Information Area:

Each control block contains an information area. The **IQ address** and **zone** number are automatically entered in the information area of each IQ component's control block when a roll call is run. The IQ address and zone can also be manually entered or changed by selecting either number and entering a new one. A valid IQ address is a number from 1 to 250. Valid zones range from 1 to 8.

Under the IQ address and zone number is a line for the **Model**, **Location**, and **Purpose**. These lines are provided for you to enter information which will help you identify the *AMB-5* controlled by the control block.

The phrase "On Line" will be displayed with green letters in the upper right corner of the information area for all *AMB-5s* which are found by the IQ software. This makes is easy to tell which control blocks are controlling active units and which are not.

DSPI:
The DSPI or Data Signal Presence Indicator is an LED on the front panel of the unit which flashes whenever it receives a data signal, addressed to it. It can be forced to stay on as an aid to troubleshooting by turning the DSPI control on. If the DSPI is forced on by the IQ software and the indicator does not light on the unit, it signifies that there is a break in communication before it or the unit is defective.

Tip: Use the DSPI light to verify that all units in the system are responding to IQ commands. For example, turn all DSPI lights on before a show to verify that all equipment is operating properly.

Tip: When there is more than one *AMB-5* in a system, use the DSPI light to quickly identify a particular one. For example, turn the DSPI light on to help a technician quickly locate a particular unit in a large equipment rack.

Aux:
The AUX port of the *AMB-5* is turned on and off with the Aux control. When turned on, the AUX port supplies +10 VDC across pins 3 (+) and 1 (gnd) of the 3-pin AUX connector located on the back panel of the unit. This can be used to signal non-IQ devices or it can be used to turn non-IQ components on and off (with the use of a solid state relay). For example, an auxiliary rack cooling system can be manually turned on with the Aux feature.

The AUX port also has the ability to sense the presence of an external signal. When a logic "high" is detected across pins 2 (+) and 1 (gnd) of the AUX connector, an **asterisk** will appear beside the Aux control to signal its presence. Normally, the AUX port is left to float. This means an asterisk may appear most of the time and can be disregarded. To use this function, the AUX port must be held "low" until a legitimate logic "high" is received.

Sel:
An *AMB-5* is selected with the Sel control. This feature is presently used by the Turbo version of the IQ-MSD software only. When switching to the graphics mode, a graphics display module will be automatically created and displayed for all selected units. The Display control of the Graphics control block must also be switched to the "Selected" position. The Sel control serves no purpose for other versions of the IQ-MSD software.

Input:
With one exception, the gain of each input is controlled separately for each output channel. The exception is Input 6 which is dedicated as an ambient sensing input for Channel 1. Its gain can be manually controlled for Channel 2 only. The range of the input level controls is +25 to –100 dB. When a gain control is turned down as far as it will go (–100 dB) the input control will say "Off".

Keyboard: The input gain can be incremented and decremented in ˚ dB steps by pressing [+] [–]. A whole number can be entered directly by first selecting the desired Input gain control and then entering a level with the number keys. Do not use [–] to enter a negative number. Instead, use the [S]. ("S" means "sign" to the IQ program.) For example, enter [S] [1] [0] to enter a level of –10 dB. A level change is not sent to the *AMB-5* until either [Enter ↵] is pressed or the cursor is moved to another control.

Mouse: Press and hold [⇧ Shift] and move the mouse (🖱) upward to increment and downward to decrement. Do not click or drag while moving the mouse.

The input gain control has its own "emergency mute" feature. Pressing [Spacebar] when an Input gain control is selected immediately mutes the gain (sets it to –100 dB). This feature is a toggle. If the control is still selected, press [Spacebar] again to toggle back to the previous setting. The previous level setting is lost if the cursor is moved to another control.

All:
The level of each output channel can be increased or decreased in 3 dB increments with the All controls. To do so, select one of the controls labelled "03" under the "All" heading and press [+] or [–] to increment or decrement it. The output level change will be reflected by an appropriate increase or decrease of each input gain setting.

Bus:
The *AMB-5* has two switchable "bus" outputs (one for each output channel). They receive the exact same signal as the main outputs. Because Channel 1 can be controlled by the algo pack, its bus output receives a post-processed audio signal. The bus outputs are each turned on and off with the Bus controls.

The purpose for the audio bus outputs is twofold: First, to provide a switchable output for each channel; Second, to make it possible to connect many mixer/multiplexers together to the same audio bus without loading down the outputs of each one. The first capability is very handy for small installations which want an on/off output for recording. The latter capability enables large number of mixer/multiplexers to be connected, for example, to an airport paging bus. The bus outputs would be off until an airport terminal gate needs to make an announcement. When this happens, the appropriate bus output of only the affected mixer/multiplexer is turned on, allowing the announcement to be sent to the paging bus.

ASA:
The ASA control affects only Channel 1. "ASA" means "Auto-System Algos" and it control serves three purposes. First, it is used to turn all *algos* on and off. Turning ASA off, restores manual-only control and causes the manual Channel 1 control settings of the *AMB-5* control block to be downloaded to the unit. Turning ASA on restores automatic operation. To turn ASA on and off, select it and press [Spacebar] just as you would for most other on/off controls. Turning the ASA control on and off can also be used to switch between two different Channel 1 mixes. See the tip at the beginning of Section 2 for more on this.

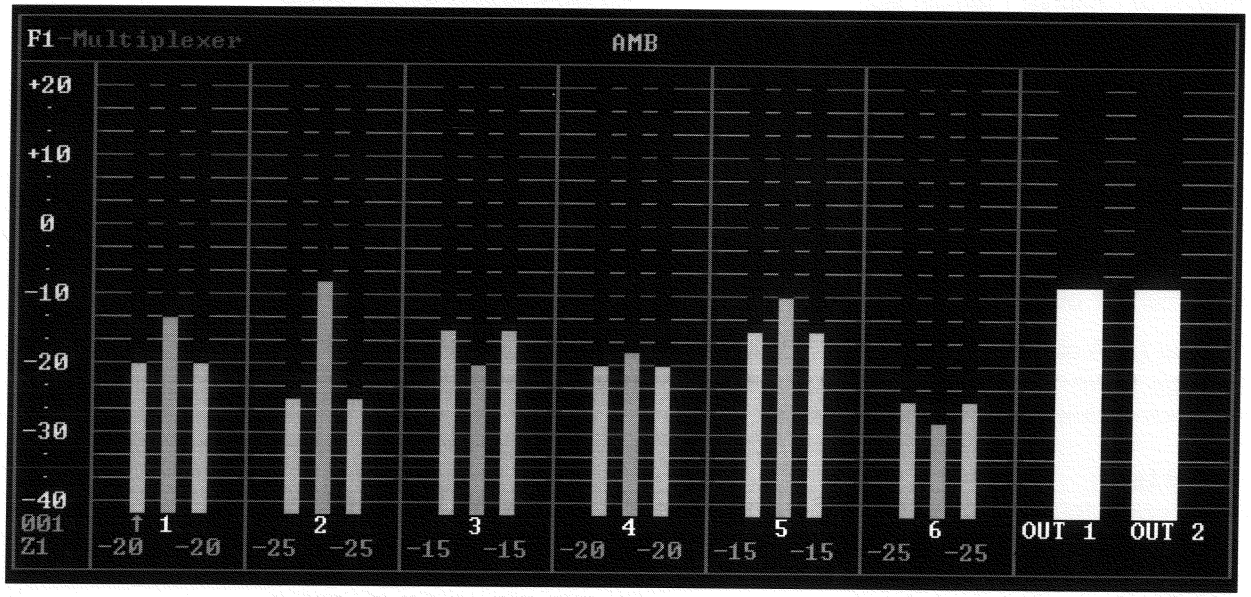
The second function of the ASA control is to serve as a door or entrance to the sub-block screens where the *algos* are configured. To bring up to the sub-block screens, highlight the ASA control and press either [Enter ↵] or [↓].

The third function of the ASA control is to tie several sensing mixer/multiplexers together in a large *IQ System*. For example, several *AMB-5s* could be wired with parallel inputs. Each *AMB-5* would be switched on as needed via the ASA control. A supervisor could use this capability to "listen in" or monitor events. It could be used to control recording or to provide additional feeds for overflow seating.

IMPORTANT: Turn ASA off before switching to the sub-block screens and making changes to the *algo* settings. The IQ software is designed for changes to be made to the *algos* with ASA off. Although it is possible to make changes while ASA is on, the algo pack is not correctly configured until the ASA control is switched from off to on. *Note: ASA can be toggled on and off from the sub-block by pressing* [Ctrl]+[A].

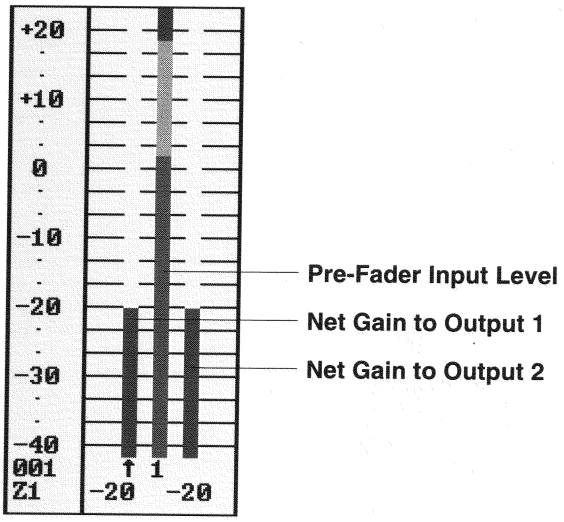
3 Display Screen

Pressing [F2] from an *AMB-5* control block or sub-block summons the Display screen of the selected *AMB-5*. Only one unit can be shown in the Display screen at a time.



The Display screen provides a dynamic representation of peak audio level of each input and output channel as well as the net gain setting of each input.

When you are ready to leave the Display screen, press [Esc], [F1] or (right mouse button) to return to the control block screen.



Input Level:
The peak input level of each audio input (1–6) is represented by a moving **multicolor bar graph** in the center of each input column on the screen. The portion of the bar that is equal to or less than +1.5 dB is **green**. The part between +1.5 dB and +18 dB is **yellow**. The part above +18 dB is **red**. Signal levels below –40 dB are not visible. The level shown is the pre-processed level (similar to a pre-fader or cue level on a mixing console). *Note: All dB levels are dBu.*

Gain:
The gain of each input is shown by the **blue bar graphs** which flank both sides of the input level bar graph. Each bar's numeric value is displayed below the bar. The gain which feeds each channel is controlled separately for each input. The gain bar graphs of Channel 1 remain stationary if ASA is turned off. If ASA is turned on the gain bar graphs of Channel 1 may move, showing the operation of the various *algos*. The Channel 1 gain bar graphs really depict the "net" gain because they show the gain settings which result from all of the *algos* working together.

With ASA off, the gain of both channels can be controlled from the Display screen with the keyboard or a mouse as described below:

Keyboard: (ASA off.) Use ← → to select the input gain you want to change. The selected input gain will have a small up arrow displayed below its bar. Use ↑ ↓ to increment or decrement the input level in ° dB steps.

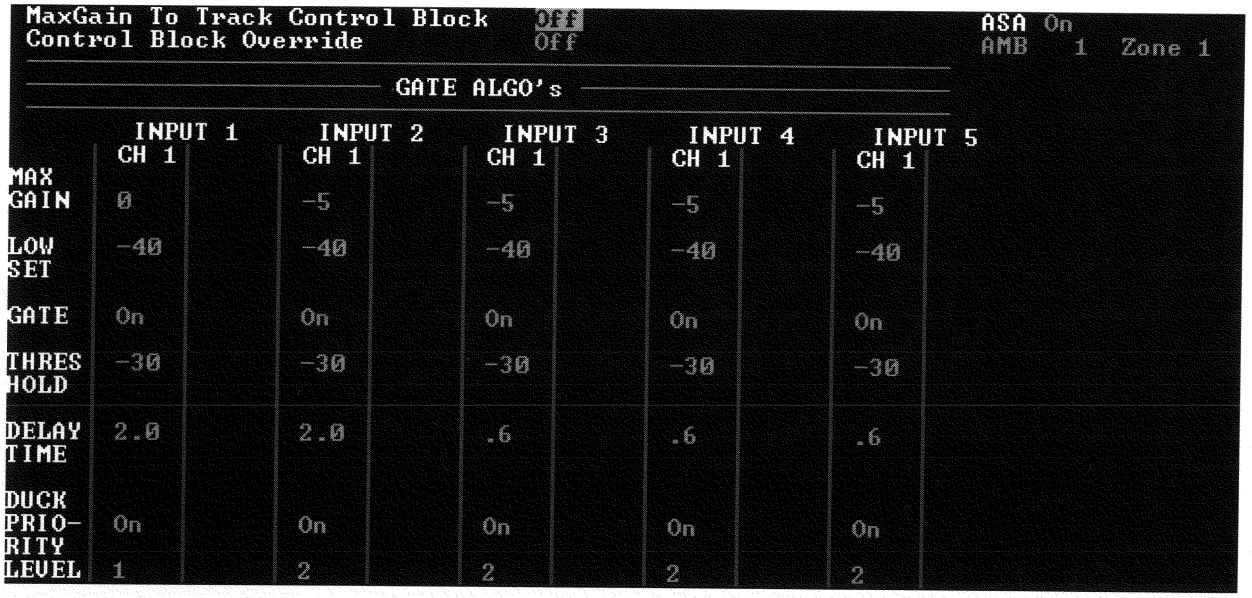
Mouse: (ASA off.) Move the mouse (🖱) sideways to select the input gain you want to change. Move the mouse (🖱) upward to increment and downward to decrement. Do not click or drag while moving the mouse.

Output Level:
The peak output level of each channel is shown by the wide **yellow bar graphs** on the right side of the Display screen.

Note: The Display screen is especially valuable when configuring the algo pack. It will be referred to several times during Section 4.

4 *Algo* Control

The real power of an *AMB-5* lies in its algo pack, the collection of self-running *algos* which control the behavior of Channel 1 when the ASA control is turned on. The *algos* are configured from the sub-block screens. To switch to the sub-block screens, select the ASA control in the control block screen and press either Enter or ↓.



The above illustration shows only the first *AMB-5* sub-block screen. There are actually five. Use PgUp PgDn to move from one sub-block screen to the next. And use Home End to move from the first sub-block screen to the last.

Unlike the *SMX-6* which has *algo* controls for both Channel 1 and 2, the *AMB-5* has *algo* controls for Channel 1 only. Also, there are no *algo* controls for Input 6 since it is dedicated as the ambient sensing input.

Max Gain To Track Control Block:
Notice the control labeled "Max Gain To Track Control Block" at the top of the first sub-block screen. When this on/off control is turned on it causes the Max Gain controls in the Gate *algo* to be set equal to the input gain settings of the control block. By copying the control block settings to the Gate *algo* Max Gain controls you can easily initialize them. See Section 4.1 for a description of the Max Gain control.

Tip: Turn on the Max Gain To Track Control Block then switch back to the control block. Set the maximum level before feedback for all microphones with the control block input gain controls. When you switch to the sub-blocks again, the Max Gain settings will be automatically adjusted to avoid feedback.

Turning on the Max Gain To Track Control Block control does not cause the control block settings to be immediately copied to the Gate *algo* Max Gain control. The action takes place only when switching from the control block screen to the sub-block screen. This means you must go back to the control block screen (press Esc or PgUp) and then return to the sub-block screen for the settings to be copied.

Remember: When Max Gain To Track Control Block is turned on, the gain settings are copied every time the switch is made from the control block to the sub-blocks. Turn this feature off when you no longer want the Max Gain settings changed.

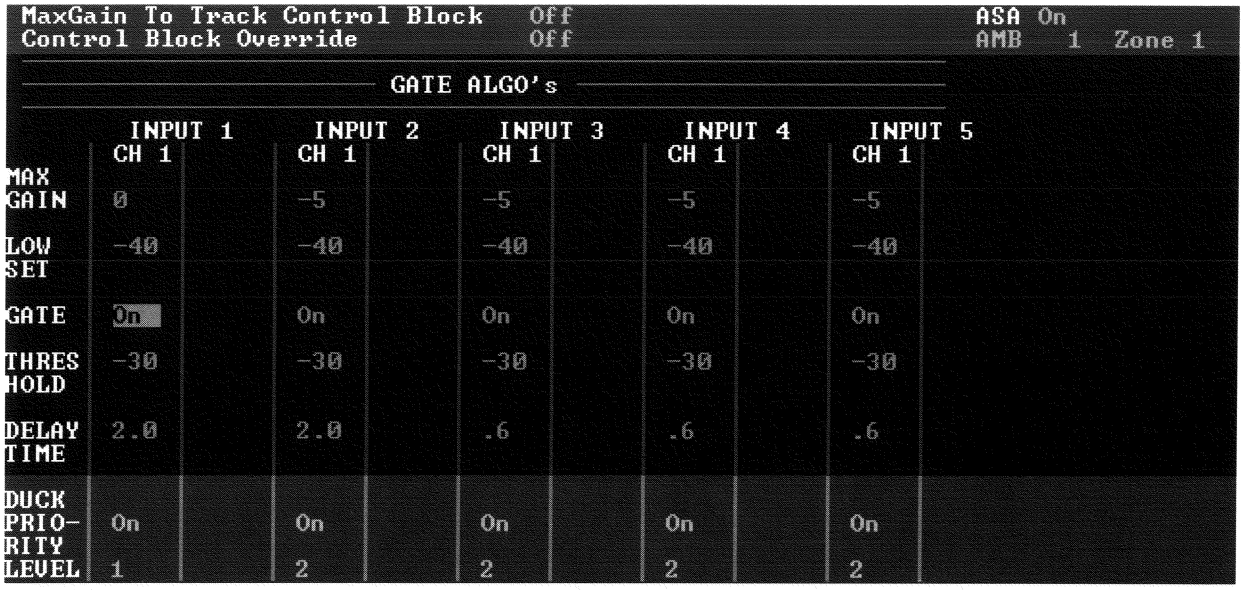
Control Block Override:
The second control at the top of the first sub-block screen is the Control Bock Override control. It is reserved for future use and should not be used now. Please leave it in the off position.

A description of each *algo* follows. They are listed in the same order as they appear in the sub-block screens.

4.1 The Gate *Algo*

"Gating" is one of the most important *AMB-5* functions. It shuts off inputs that aren't in use. Gating is often used to prevent feedback. If several mics in a system are on at the same time, the system's gain could become so high that it rings with transients, or breaks into feedback.

Lowering the overall gain is one way to fix this problem, but some mic levels might become too low. Gating is a better solution. A microphone is "live" only when a sound is loud enough to open its gate (for example, speaking, singing, etc.).



Max Gain:
The Max Gain control sets the maximum gain for an input. The gain can't rise higher, even when the Auto level *algo* is active. The Max Gain control also sets the "gate open" gain when the Gate *algo* is in use.

When ASA is on, the Max Gain settings in the sub-block override the corresponding control block gain settings. They work like the control block gain settings when most of the *algos* are off. When ASA is off, the control block gain settings take over.

The Max Gain control has its own "emergency mute" feature. Pressing Spacebar when a Max Gain control is selected immediately mutes the gain (sets it to –100 dB). This feature is a toggle. If the control is still selected, press Spacebar again to toggle back to the previous setting. The previous level setting is lost if the cursor is moved to another control.

Tip: The final gain settings are usually made by listening. If you set the gain too high, the system could break into feedback. It is possible to attenuate all channels at once

using the secondary mix capability of the control blocks. First, set the control block levels far below the feedback point. Now ASA can be used to activate the control block mix which will pull all inputs below the feedback level. (The Max Gain To Track Control Block control should be turned off.) The ASA control can be toggled on and off from the sub-block screens by pressing Ctrl+A.

Note: Toggling ASA on and off switches between the sub-block mix and the control block mix. This gives you two different mixes only if Max Gain To Track Control Block is not turned on.

Low Set:
The Low Set gain is the "gate closed" gain. Low Set is off by default (about –100 dB). Set it higher if you want the signal to be audible even when the gate is "closed."

Gate:
The Gate *algo* is turned on and off with the Gate control. Turning the Gate control off also turns off the Duck Priority and Gate Count *algos.*

Threshold:
The Threshold control determines when an input gate opens. When a signal exceeds the Threshold, the gate opens causing the gain to increase from the Low Set value to the Max Gain value. When a signal drops to the Threshold or below, the gate closes and the gain drops from Max Gain to the Low Set value. *Note: If the Threshold is set too high, the gate will never open, or only open sporadically. If it's too low, background noise may keep the gate open all the time.*

When a signal exceeds the Threshold, the gate opens almost instantly. The optimum Threshold setting depends on the signal. Speech varies more rapidly than music, and often needs a lower Threshold to keep the gate open. For a microphone, the Threshold can be set a few decibels above a room's ambient sound for ambience rejection, or set higher to reject incoming far-field or off-axis sounds.

Tip: The Display screen (press F2) is very helpful when setting the Threshold controls. It displays the input signal level, making it a simple matter to select an appropriate Threshold level. For example, it is quickly apparent from the Display screen that the gate will not open if the Threshold control is set at a higher level than the input signal.

Delay Time:
The Delay Time control determines how long the gate will stay open after the incoming signal drops below the Threshold. The optimum Delay Time setting depends on how rapidly the input changes. If a speaker walks slowly from one mic to another, you will probably want to use a long Delay Time for each mic to prevent an audibly abrupt cutoff. At a conference however, when debaters compete for microphones, a shorter Delay Time would probably be more appropriate.

The following list shows the possible settings for each Gate *algo* control. *Note: All dB settings are dBu.*

Max Gain: Off; –99½ dB to +25 dB in ½ dB steps

Low Set: Off; –99½ dB to +25 dB in ½ dB steps

Gate: On/Off

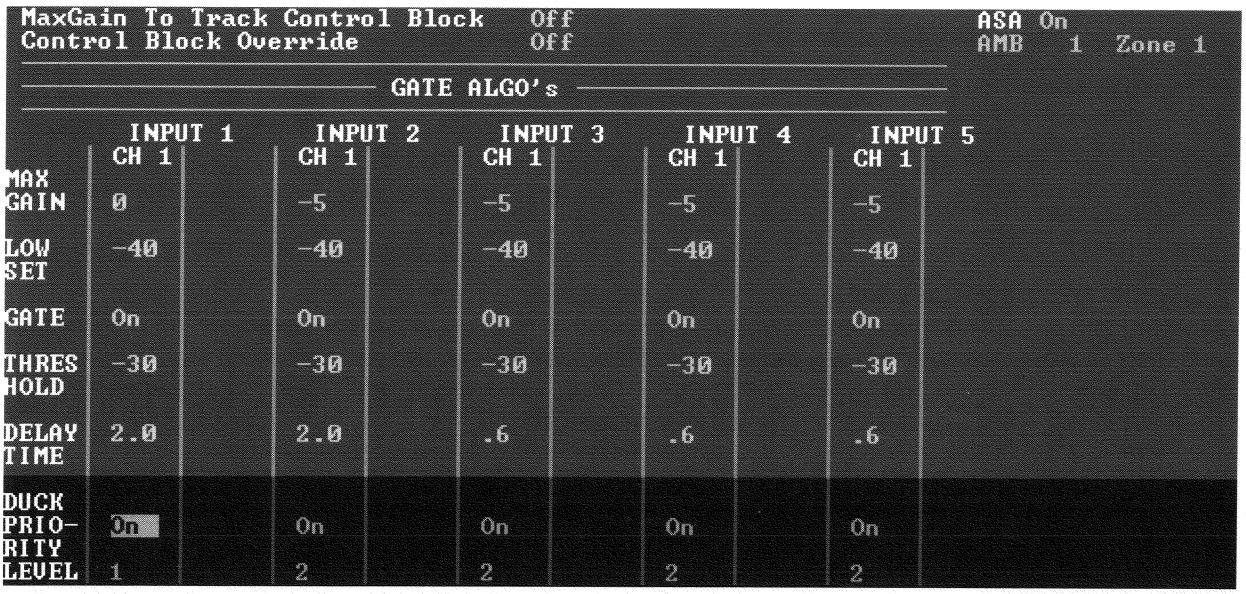
Threshold: Off; –99½ dB to +25 dB in ½ dB steps

Delay Time: 0.2, 0.4, 0.6, 0.8, 1, 1.5, 2, 3, 4, 6, 8, 10, 12, 15, 20, 30 seconds

4.2 The Duck Priority *Algo*

"Ducking" is the attenuation of an input when another input with a higher priority is activated by the Gate *algo*. When the higher priority gate opens, all inputs with lower priorities drop to their Low Set values. To activate the Duck Priority *algo*, turn it on and set the priority level.

IMPORTANT: The Duck Priority *algo* is dependent on the Gate *algo*. Both the Gate *algo* and the Duck Priority *algo* must be on for Ducking to function.



Duck Priority:
The Duck Priority *algo* is turned on and off with this control.

Duck Priority Level:
The Duck Priority Level control determines which input takes precedence. The priority can be set from 1 to 6 with 1 as the highest priority and 6 as the lowest. A separate priority setting is provided for each channel, making it possible to give an input a different priority for each output channel. The Duck Priority Level control is ignored if either the Duck Priority or Gate controls are turned off.

If only two inputs are used, there only needs to be two priority levels (perhaps 1 and 2). If you are unsure how many priority levels will be needed or what the final priorities will be, start with levels 3 and 4. The priorities above and below remain available for assignment to other inputs.

Tip: When a signal ducks, the gain drops to the Low Set level for that input. The default value of Low Set is off (–100 dB). This is such a large transition that it may sound abrupt. Pick a higher value for Low Set to avoid this problem if a smoother transition is desired. It is possible to set the Low Set control so high that the ducked signal is still audible over the background noise. This may be desirable when mixing a debate or conference.

Note: The Delay Time control of the Gate algo also affects the Duck Priority algo. When a high-priority input falls below its gate Threshold, all ducked inputs must remain at their Low Set levels until after the Delay Time.

The following list shows the possible settings for each Duck Priority *algo* control.

Duck Priority: On/Off

Duck Priority Level: 1, 2, 3, 4, 5, 6

4.3 The Input Compressor/Limiter *Algo*

The Max Gain settings you select in the Gate *algo* should be acceptable for consistent input levels—but sometimes the unexpected happens and an extremely loud input signal is present. The Input Compressor/Limiter *algo* "reins in" input signals that suddenly become too loud. If the excessive loudness of the input signal is not transitory but is sustained, use the Auto level *algo* to control it (Section 4.4). The Input Compressor/Limiter *algo* is located on the second *AMB-5* sub-block screen.

Compress:
To turn the compressor on, select the Compress control for the desired input and press Spacebar.

Threshold:
The Threshold control sets the level at which compression begins. A signal at this level or below is not affected, and a signal that goes above this level is compressed.

Ratio:
The Ratio control sets the amount of compression applied for every step over the Threshold. This is commonly referred to as a "compression ratio". For example, when a 4:1 compression ratio is applied to a signal that exceeds the Threshold by 20 dB, the signal is attenuated 15 dB. When an infinite compression ratio (∞:1) is chosen, the compressor works like a limiter and prevents the level from exceeding the Threshold. *Note: A 1:1 compression ratio will not affect an incoming signal.*

Release Time:
Any input level increase over the Threshold is compressed almost instantly. The Release Time *Algo* controls the time it takes for the gain to return to normal when the level drops below the Threshold. Short release times allow the compressor to respond to rapid changes in input level.

Compression (Attack Time):
The Compression control can be set either "Fast" or "Slow" to control the attack time of the Input Compressor/Limiter *algo*.

Set the Compression control to "**Fast**" to cause the *AMB-5* to have the shortest attack time possible. This will be very fast—so fast that the compressor can stop extremely sudden loud transients like the sound caused by a dropped microphone. This is a good setting for rapid speech.

Set the Compression control to "**Slow**" for more gentle sound sources such as most music. When the "Slow" setting is used, the Release Time cannot be set any less than 0.4 seconds to avoid noticeable pumping of the Input Compressor/Limiter *algo*.

The following list shows the possible settings for each Input Compressor/Limiter *algo* control. *Note: All dB settings are dBu.*

Compress: On/Off

Threshold: Off; –99½ dB to +25 dB in ½ dB steps

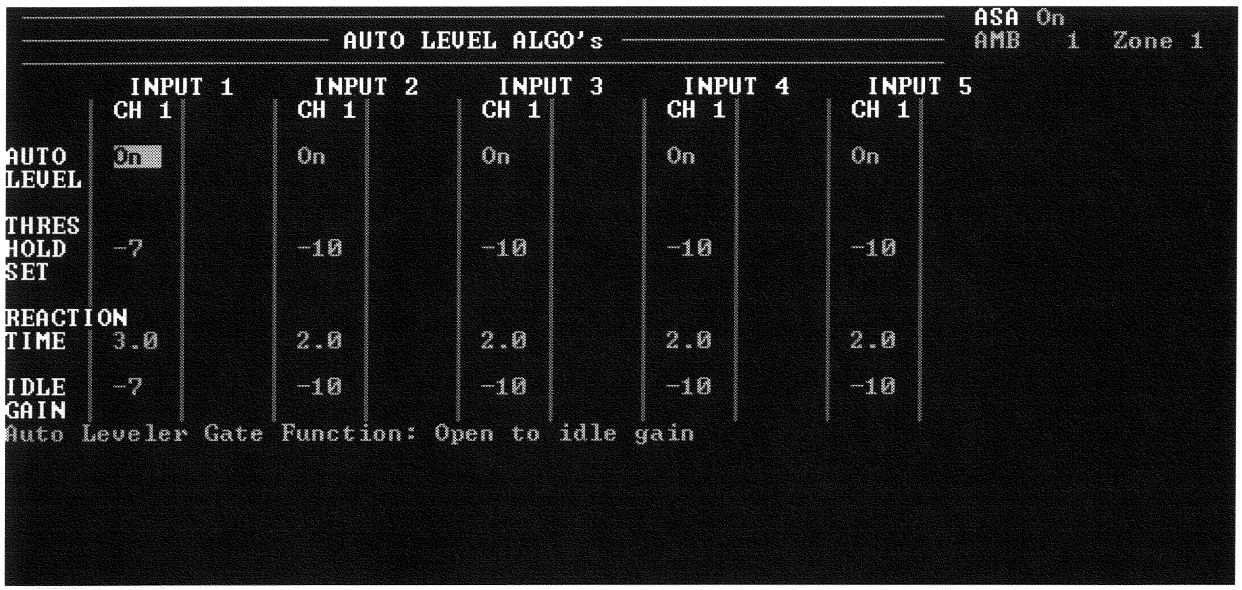
Ratio: 1:1, 2:1, 4:1, 8:1, 16:1, 32:1, ∞:1

Release Time: 0.2 (Fast compression only), 0.4, 0.6, 0.8, 1, 1.5, 2, 3, 4, 6, 8, 10, 12, 15, 20, 30 seconds

Compression: Fast/Slow

4.4 The Auto Level *Algo*

The Auto Level *algo* is located on the third *AMB-5* sub-block screen. Unlike the Input Compressor/Limiter *algo* which is designed to handle sudden or transient jumps in input level, the Auto Level *algo* is designed to control long-term changes in signal level. Using it, the average output level can be kept relatively constant over a longer time interval while transients are allowed to pass. This feature creates a more natural sound with increased dynamic range.



Auto Level:
To turn the auto level feature on, select the Auto Level control for the desired input and press Spacebar.

Threshold Set:
The Threshold Set control sets the "target" signal level for the Auto Level *algo*. If Threshold Set is –20 dB, and the output signal level is –20 dB, nothing changes. The signal passes at the same level. But if the signal is above –20 dB, the Auto Level *algo* will reduce the gain to bring the level down to –20 dB. And if the signal is below –20 dB, the gain is brought up to that level.

Reaction Time:
The Auto Level *algo* continually adjusts the signal level toward the Threshold Set value. But it doesn't adjust the signal in one jump. Instead, the Reaction Time control sets the size of each step. For example, if the audio input signal stays at a constant level below the Threshold Set value, the gain would increase it to the Threshold Set level over the time (in seconds) specified by the Reaction Time control. In the "real world," audio signals continually change, and so does Auto Level's step size. This approach produces a very natural sounding auto level control.

A Reaction Time of 4 seconds is a good starting point for speech. Try 6–10 seconds for music. Although a 4 second Reaction Time might seem slow, remember that the Auto Level *algo* compensates for average differences in level—such as between soft-spoken and boisterous speakers. If a sound source occasionally "blasts," you might also use the Input Compressor/Limiter *algo* to control the transients.

Tip: The Auto Level *algo* never raises the gain higher than Max Gain. As you set the value of Max Gain higher, the Auto Level *algo* will have more room to adjust the incoming signal. For inputs with an unusually wide dynamic range, set Max Gain as high as possible so the Auto Level *algo* can bring up the quietest passages.

Tip: Use a long Reaction Time when the system is used to mute background music during a page. This provides a pleasing and smooth transition back to background music after the page has been completed.

Idle Gain:
If the Gate and Auto Level *algos* are used together, their interaction may produce an unexpected side effect. When Max Gain is set above the Threshold Set of the Auto Level *algo*, the Auto Level *algo* could start lowering the gain every time the gate opens. If there is a great difference between the signal level and Threshold Set, the shift in gain will be very noticeable. The sound would start out loud and then fade.

The Idle Gain control can compensate for this situation. Idle Gain controls the initial gain when the gate first opens. You can think of it as the "starting gain" because it overrides the Max Gain when the gate opens. Start with Idle Gain equal to Max Gain. If the level audibly drops each time the gate opens, try a lower value for Idle Gain.

IMPORTANT: The Idle Gain *algo* functions only when three conditions are met. 1) The Gate *algo* is turned on. 2) The Auto Level *algo* is turned on. 3) The Auto Leveler Gate Function control is set to "Open to idle gain". See the Auto Leveler Gate Function control below for more information.

IMPORTANT: Never set Idle Gain higher than Max Gain. Because the Idle Gain overrides the Max Gain, it is possible to force the gain above the Max Gain setting.

Auto Leveler Gate Function:
The purpose of the Auto Leveler Gate Function control is to help create smoother transitions when both the Gate and Auto Level *algos* are operating. Both of these *algos* must be operating for this feature to function.

The "**Open to idle gain**" setting is best for systems with relatively constant sound levels as with background music. When an input gate is opened, its gain is set to the Idle Gain setting. See the description of the Idle Gain control for more information.

The "**Open to last position**" setting is best for systems with widely varying sound levels where an Idle Gain setting is difficult to choose. When an input gate opens, the Idle Gain setting is ignored and the gain is restored to the previous level it had just prior to its closing. An example of when this setting is very useful is when speakers with very different voice levels share the same microphone.

The following list shows the possible settings for each Auto Level *algo* control. *Note: All dB settings are dBu.*

Auto Level: On/Off

Threshold Set: Off; –99½ dB to +25 dB in ½ dB steps

Reaction Time: 0.2, 0.4, 0.6, 0.8, 1, 1.5, 2, 3, 4, 6, 8, 10, 12, 15, 20, 30 seconds

Idle Gain: Off; –99½ dB to +25 dB in ½ dB steps

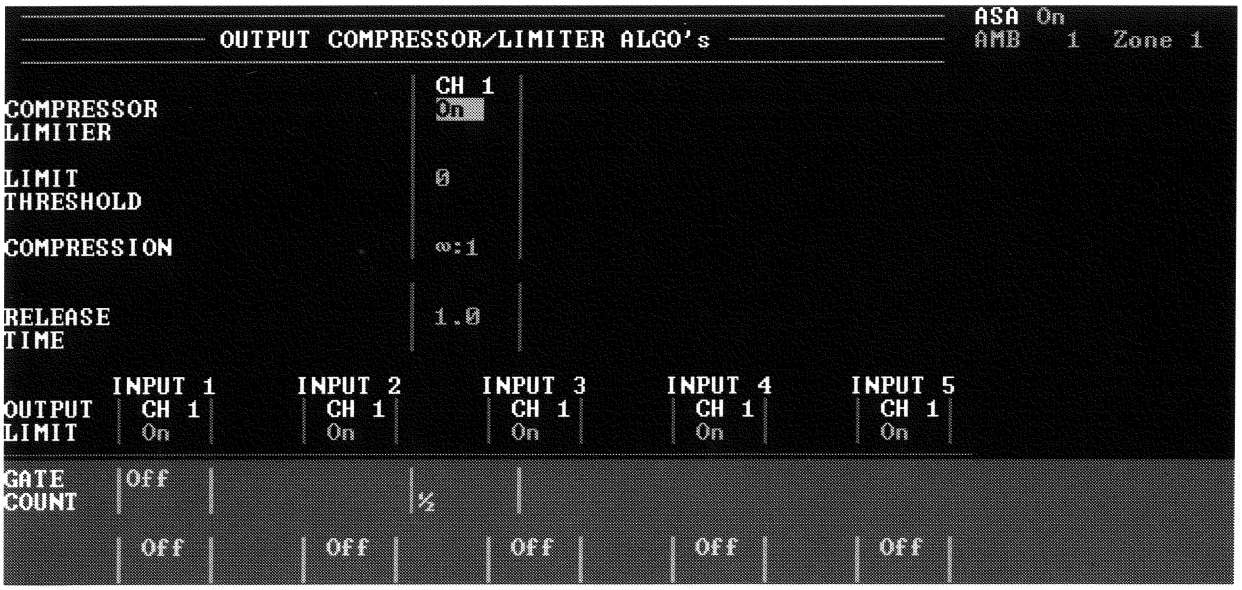
Auto Leveler Gate Function: Open to idle gain/Open to last position

4.5 The Output Compressor/Limiter *Algo*

Even if every input is compressed, there is still a chance that the mix might become too loud. The Output Compressor/Limiter *algo* keeps the output from exceeding a predetermined level. It's especially valuable when used as an output limiter for the protection of amplifiers, loudspeakers and other audio equipment.

The Output Compressor/Limiter *algo* is located on the fourth *AMB-5* sub-block screen. It works in a very interesting way. Instead of compressing the signal on the mixing bus, it compresses the signal at each input. Compression occurs only when the mix exceeds the Limit Threshold. It doesn't matter how loud a particular input gets, the mix which feeds each output channel must exceed the Limit Threshold before compression occurs.

The controls that configure the Output Compressor/Limiter *algo* are very similar to those of the Input Compressor/Limiter *algo*.



Compressor/Limiter:
To turn the output compressor/limiter feature on, select the Compress/Limiter control for the desired output channel and press Spacebar.

Limit Threshold:
The Limit Threshold control sets the peak signal level above which compression occurs. If the *AMB-5* is connected to audio equipment that has a specified maximum input voltage, it is easy to limit the *AMB-5's* output to that level. If the specified input level were 0 dBu (0.775 V), the Limit Threshold could be set to 0 and the Compression Ratio set to ∞:1. In practice, this is a little harsh. A Limit Threshold of –2 dBu with a Compression Ratio of 16:1 will attenuate an incoming signal 32 dB before the output level reaches 0 dBu. When used with the Input Compressor/Limiter, the Output Compressor/Limiter provides some of the most accurate and predictable audio system protection available today.

Tip: If you're not sure where to set the Limit Threshold, press F2 to view the bar graph display. The yellow peak output level bars can be helpful.

Compression Ratio:
The Compression Ratio control sets the amount of compression applied for every step over the Limit Threshold. For example, when a 4:1 compression ratio is applied to a signal that exceeds the Limit Threshold by 20 dB, the signal is attenuated 15 dB. When an infinite compression ratio (∞:1) is chosen, the compressor works like a limiter and prevents the output level from exceeding the Limit Threshold. *Note: A 1:1 compression ratio will have no affect on the signal.*

Release Time:
As with the Input Compressor/Limiter, the Output Compressor/Limiter *algo* responds almost instantly to signals above the Threshold. The Release Time control sets the time for the gain to return to normal after the signal falls below the Limit Threshold.

Output Limit:
The Output Limit control lets you select which inputs are compressed, and which are not. This is a highly flexible feature, and can be used in some very advanced setups. To make the Output Compressor/Limiter work like a conventional output compressor, make certain that all of the Output Limit controls are turned on. Selectively turning off an Output Limit control enables its input to push all remaining inputs down whenever it exceeds the Threshold. This is a specialized function and is offered only for those unique circumstances which require it.

Tip: You don't have to compress all inputs. You could restrict compression to an offending input, leaving unchanged the signals that are unlikely to exceed the Limit Threshold. The Output Compressor/Limiter *algo* allows the *AMB-5* to maintain a more natural sense of dynamics, with fewer audible side-effects.

The following list shows the possible settings for each Output Compressor/Limiter *algo* control. *Note: All dB settings are dBu.*

Compressor/Limiter: On/Off

Limit Threshold: Off; –99½ dB to +25 dB in ½ dB steps

Compression Ratio: 1:1, 2:1, 4:1, 8:1, 16:1, 32:1, ∞:1

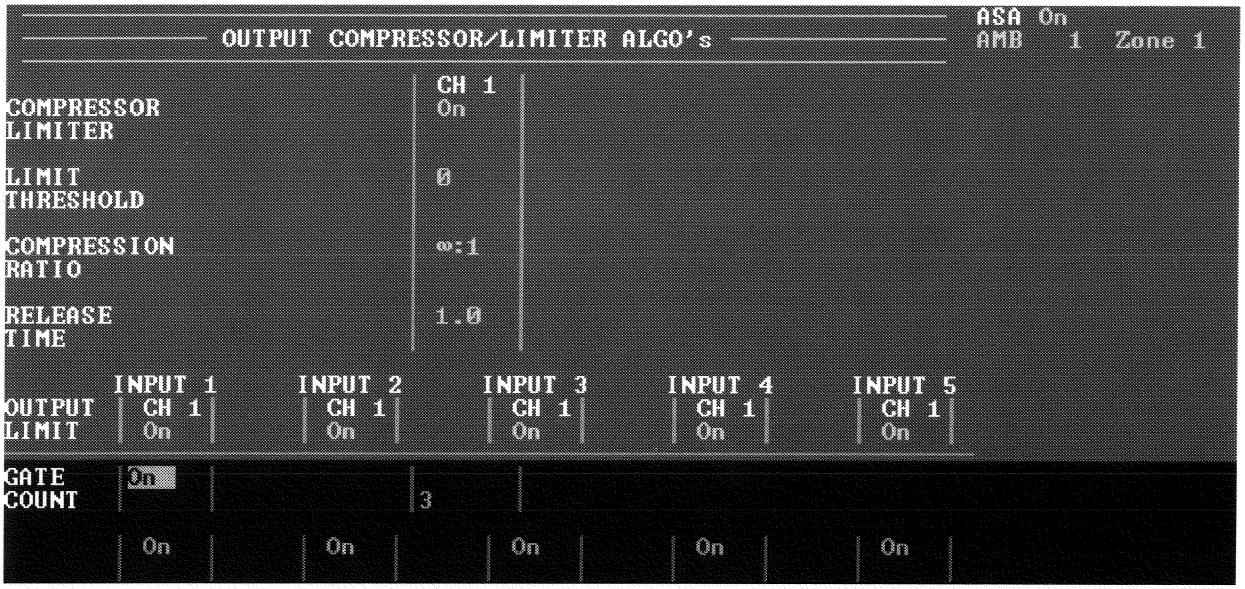
Release Time: 0.2, 0.4, 0.6, 0.8, 1, 1.5, 2, 3, 4, 6, 8, 10, 12, 15, 20, 30 seconds

Output Limit: On/Off

4.6 The Gate Count *Algo*

The Gate Count *algo* is located on the fourth *AMB-5* sub-block screen. It works in conjunction with the Gate *algo*. The Gate Count *algo* appears below the Output Compressor/Limiter *algo*.

The Gate Count *algo* keeps track of how many gates are open at a time, and provides feedback control when a large number of mics are open. A similar feature of other manufacturers' products is called "NOM" (Number of Open Mics). When only two or three mics are open, system gain is usually too low to cause feedback. But as more mics open, the system gain can increase enough to cause feedback.



Gate Count:
Immediately to the right of the Gate Count label is the master on/off control. Turn this control on to activate the Gate Count *algo*.

IMPORTANT: The Gate Count *algo* is dependent on the Gate *algo*. Both the Gate *algo* and the Gate Count *algo* must be on for the latter to function.

Notice that each input also has an individual on/off control. They are located across the bottom of the screen and are used to control which inputs participate in the Gate Count function. Inputs with this control turned off are not counted.

Count Level:
The Count Level control determines how much each input is attenuated as more gates open. It is entered in dB. When only one mic in a count group is active, there is no added attenuation. Each additional open mic increases the attenuation by the "Count Level". For example, if the Count Level were 3 dB and five mics were open, the attenuation would be 12 dB, or (NOM – 1) x Count Level.

The following list shows the possible settings for each Gate Count *algo* control.

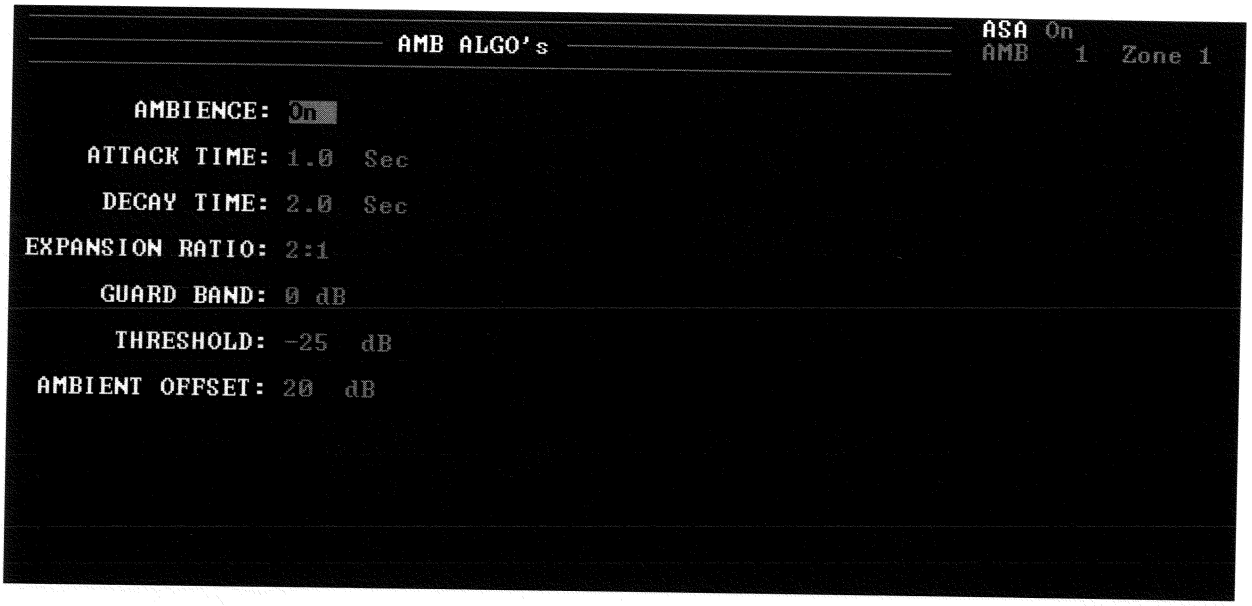
Gate Count: On/Off

Count Level: ½ to 3 dB in ½ dB steps; 3 to 12 dB in 1 dB steps; 16 dB

4.7 The AMB *Algo*

The AMB *algo* is unique to the *AMB-5*. It controls the ambient sensing feature of the unit and is located on the last (fifth) *AMB-5* sub-block screen.

The AMB *algo* monitors the ambient sense input (Input 6) and automatically adjusts the mix output level of Channel 1 up or down accordingly. This enables the *AMB-5* to automatically mix and control the overall loudness in an environment with background noise levels that vary. For example, it can insure that a factory paging system will never be too quiet or too loud when noisy machinery turns on and off.



Functionally, the AMB *algo* operates after all other *algos* have processed the audio signals. You can think of the AMB *algo* as a "black box" which sits after all other *algos* in the *AMB-5* algo pack.

There are external factors that can greatly affect how well this *algo* functions. The two most important ones are the location of the ambient sense microphone and the location of the loudspeakers. The ambient sense microphone must be located in a position where it can pick up the ambient noise level without also picking up unwanted sounds. The ambient sense microphone must also be located far enough away from the loudspeakers so that they do not inadvertently trigger the AMB *algo*. It may take some experimentation to find both the optimum location for the ambient sense microphone and the optimum settings for this *algo*.

Tip: Unwanted sounds can often be filtered out with a band-pass filter or by using an ambient sense microphone with a selective pass band. For example, a 500–4,000 Hz band-pass filter could be added so the sense input responds only to the noise which would interfere with speech intelligibility. Also, a high-pass filter could be added so the sense input does not pick up low-frequency air conditioning noise (which can sometimes be very loud).

Tip: Shield the ambient sense microphone from excessive wind noise so that it does not prematurely activate the AMB *algo*.

WARNING: The AMB *algo* bases its operation on the Gate *algo* Max Gain settings for each input. **It is extremely important that the Max Gain settings be set properly to prevent excessive levels which could cause feedback.**

Several of the AMB *algo* controls interact with each other. For this reason we recommend that they be configured in the order shown below:

1. Double-check the Gate *algo* Max Gain settings of each input to be sure they are set properly to limited all input gains to safe levels.
2. Set the Ambient Offset Control to a desired level below the Max Gain settings.
3. Set the Threshold of the ambient sense input to an appropriate level so that the AMB *algo* begins to operate at an appropriate time. This may require some experimentation.
4. Set the Expansion Ratio so that the dynamic range of the ambience level does not exceed the available headroom of the system.
5. Set the Attack and Decay Time controls to appropriate speeds.
6. Set the Guard Band to the desired offset below the Threshold (if desired). Normally, this can be turned off by setting it to 0 (zero).
7. Turn on the AMB *algo* with the Ambience control.

Next, the controls are described in the order in which they are listed on the sub-block screen.

Ambience:
The AMB *algo* is turned on and off with the Ambience control.

Attack Time:
The Attack Time control determines how rapidly the AMB *algo* will begin to increase the output level of Channel 1 above the Ambient Offset level. Slower Attack Times cause the AMB *algo* to respond slower to increases in ambient level and prevent it from responding to sudden increases in level.

Decay Time:
The Decay Time control determines how rapidly the AMB *algo* will begin to reduce the output level of Channel 1 while the ambient sense input level is dropping. Slower Decay Times cause the AMB *algo* to respond slower to decreases in ambient level.

Expansion Ratio:
The Expansion Ratio control determines how much the gain of output Channel 1 will increase for every increase in the ambient signal level above the Threshold setting. A 4:1 Expansion Ratio setting will raise the output gain 1 dB for every 4 dB that the ambient signal increases.

Most of the time the Expansion Ratio can be set to 1:1 so that the output level increases 1 dB for every increase in the ambient sense level. However, if the ambient noise level has an extremely wide dynamic range it is possible to exceed the headroom of the system. This is when you will want to use the Expansion Ratio control to reduce the dynamic range of the AMB *algo*.

Guard Band:
The Guard Band control is an offset to the Threshold control. For example, if the Guard Band is set to 3 dB, the threshold will be 3 dB below the Threshold control setting. It is provided as a tool for making temporary adjustments to the Threshold level during setup. Normally, it should be set to 0 (zero) so that the Threshold control setting accurately represents the actual threshold.

Threshold:
The Threshold control determines at what ambient sound level the AMB *algo* will begin to operate. It is the sound pressure level (SPL) of the ambient sensing input (Input 6). When the signal level of Input 6 exceeds the Threshold setting, the AMB *algo* begins to increase the gain of Channel 1 at the rate dictated by the Expansion Ratio control.

Ambient Offset:
The Ambient Offset is like a fixed insertion loss. It is the level below the Gate *algo* Max Gain settings to which the Channel 1 gain will be set until the ambience level exceeds the Threshold setting. As the ambience level increases above the Threshold setting, the Channel 1 gain will be increased above the Ambient Offset according to the Expansion Ratio setting. The Max Gain settings always limit the maximum possible gain which any input can receive.

Inputs whose net gain is forced below the Ambient Offset by the other *algos* are ignored by the AMB *algo*. This prevents the AMB *algo* from attempting to increase the gain of inputs whose gates have been closed.

The following list shows the possible settings for each AMB *algo* control. *Note: All dB settings are dBu.*

Ambience: On/Off

Attack Time: 0.2, 0.4, 0.6, 0.8, 1, 1.5, 2, 3, 4, 6, 8, 10, 12, 15, 20, 30 seconds

Decay Time: 0.2, 0.4, 0.6, 0.8, 1, 1.5, 2, 3, 4, 6, 8, 10, 12, 15, 20, 30 seconds

Expansion Ratio: 1:1, 1.5:1, 2:1, 2.5:1, 3:1, 4:1, 5:1

Guard Band: 0, 1, 2, 3, 4, 5 dB

Threshold: Off; –99½ dB to +25 dB in ½ dB steps

Ambient Offset: 0 to 40 dB in ½ dB steps

5 *Algo* Sample Applications

The following examples show how to use the *AMB-5* algo pack in "real-world" applications—the kinds of situations commonly encountered in audio systems in the field. The *AMB-5* has the flexibility to handle a very wide range of setups.

The *algo* samples included in this section focus on how the various *algos* interact with each other and how to set appropriate values for their individual controls. Every *algo* is covered. Don't hesitate to experiment as you read along. When you've finished reviewing the examples, you should be ready to tackle your own setups.

Please note: The algo settings shown here are only given as examples. They have been selected to show the appropriate settings for a specific system and are not intended to be absolute or "ideal" values for every situation. System requirements and your own listening tests should be used to determine your optimum settings.

Subway train noise makes sound system design challenging, especially if the acoustical space is underground. With ordinary subway sound systems, announcements are often impossible to hear during train arrivals and departures. And if paging levels are set high enough to hear over the noise, they would seem unpleasantly loud when trains are away from the station.

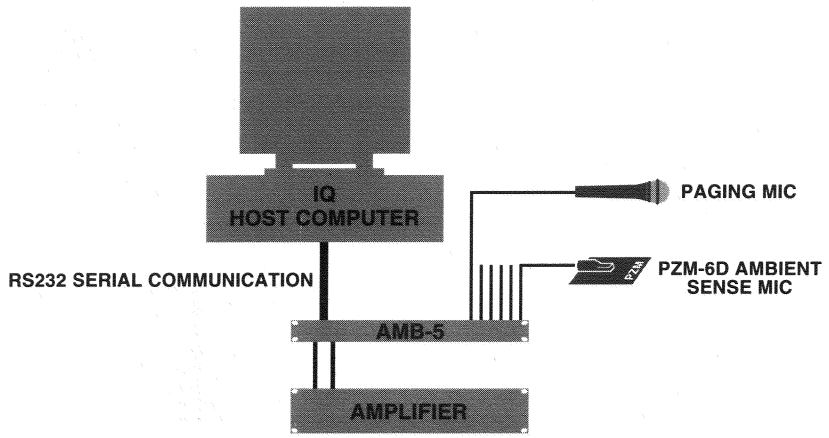
As you can see, a subway station is an ideal application for the *AMB-5*. The following example includes two different designs and two different approaches to system setup. Later in this section, a factory sound system that uses multiple ambient mics will also be described.

5.1 Subway *IQ System* Design 1

A simple *AMB-5* subway sound system might require a single ambient input for train noise and a single input for paging. Crown's *PZM*® boundary microphones work well for ambient input. For this sound system, a PZM-6D is used because it has a broad 20 Hz–20 KHz frequency response that will pick up the low frequencies of the train. The mic is placed a short distance up the tunnel so it will pick up the ambient train noise while remaining isolated from the loudspeakers of the public address system.

A single ambient microphone can be used for this job because the subway station isn't particularly large and the *PZM* mic picks up the train noise even after the train leaves the tunnel. Additional ambient mics might be required if the subway platform were much larger, or if ambient adjustments to crowd noise were required.

Subway System



CAUTION: Always remember to give careful consideration to the placement of any microphone that drives the *AMB-5's* ambient channel. As the volume of the sound system increases, the distance between the ambient mic and the loudspeakers must also be increased. If a loudspeaker under *AMB-5* control feeds into the ambient mic, you might experience the following: (1) "ambience feedback" where the *AMB-5* drives its own gain up to the maximum level; or (2) the ambient input improperly controls the gain in more subtle ways. If the ambient mic "couples" with the loudspeaker output, the system might experience a runaway gain increase that could damage system components—**but this can only happen if the maximum level (Max Gain) isn't set correctly**. Max Gain settings that protect the system and good microphone placement are the most critical aspects of an *AMB-5* installation.

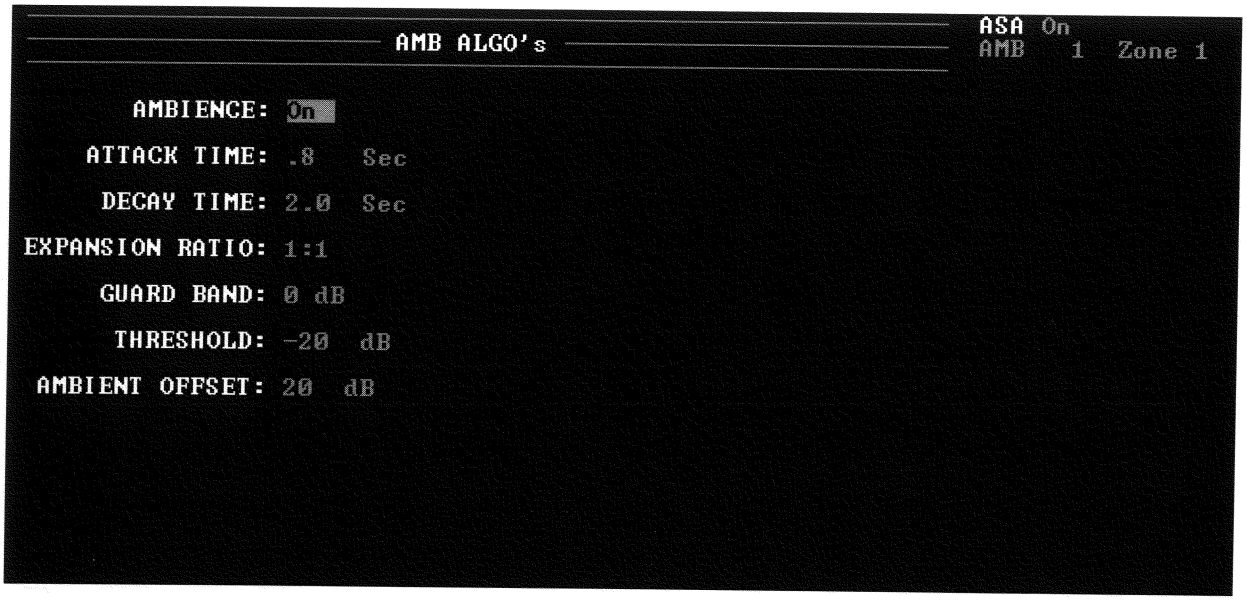
5.1.1 Setup "On the Fly"

The fastest way to set up an *AMB-5* is to do it "on the fly" (a more scientific approach is described later). System setup "on the fly" means you change settings and listen to the results until the system performs satisfactorily. To speed up this process, you can use the information provided by the bar meters on the Display screen that is accessible by pressing F2.

To set up a system "on the fly," you should first do the following: (1) select a good location for the ambient mic; (2) design the sound system to deliver uniform sound pressure levels (SPLs) to the listening area; and (3) make sure the system can deliver the maximum required SPL.

IMPORTANT: Always set up the Gate, Duck Priority, Input Compressor/Limiter, Output Compressor/Limiter, Auto-Level and Gate Count *algos* before configuring the AMB *algo* settings. If these *algos* are not configured first, it might be impossible to effectively configure the AMB *algo* settings. Configuring the AMB *algo* after all other *algos* is important because it works as an independent module or "black box" that's located down-line from the other *algos*.

The first step in configuring the AMB *algo* is to set its dynamic range. The dynamic range is defined by the highest and lowest output level. The Gate *algo* **Max Gain** settings determine the highest level. The **Ambient Offset** control determines the lower limit—as long as other *algos* don't force the gain lower.



IMPORTANT: The AMB *algo* ignores all inputs whose net gain is forced below the Ambient Offset by the other *algos*. This prevents the AMB *algo* from attempting to raise the gain of an input whose gate is closed.

When the signal level of the ambient sense input rises above the Threshold setting, the AMB *algo* will increase the Channel 1 output level above the Ambient Offset level at the rate specified by the Expansion Ratio control. The Ambient Offset is an offset below the Gate *algo* Max Gain settings. To find what the minimum gain of the AMB *algo* is, subtract the Ambient Offset from the Gate *algo* Max Gain of each input.

After the Max Gain controls and the Ambient Offset control have been set, it is time to set the Threshold. In this subway example, the **Threshold** determines the SPL in the tunnel that is required to activate the AMB *algo*. The Threshold control must be set before setting the other AMB *algo* controls.. In this example, the Threshold should be (1) above the average crowd noise, and (2) below the ambience level of the train when it's stopped at the station's platform. With the Threshold in this range, the Ambient Offset will control normal operation, and the extra noise of the train will cause an ambience adjustment. Unusually loud crowd noise might also activate an adjustment, but this is appropriate for such noise levels.

It's best to set the Expansion Ratio next because it affects the operation of the Attack and Decay Times. It can be easily set with a little experimentation. When the ambient input level rises above the Threshold, the **Expansion Ratio** determines how much the system gain will increase for every increase in ambience. A 4:1 Expansion Ratio, for example, means the ambient input will rise 4 dB before allowing the system gain to rise 1 dB. This setting determines the potential for system gain increases based on ambience control.

If a 1:1 Expansion Ratio might exceed the headroom of the system, you'll need to increase the Expansion Ratio. If on the other hand, a 1:1 Expansion Ratio does not provide enough gain, you may need to lower the Threshold setting. First check to be sure that long Attack or Decay Times aren't simply delaying an adequate gain adjustment. By decreasing the Threshold, the same peaks in ambience will represent more steps over the Threshold, and therefore provide greater potential for system gain. The hardware gain potentiometer of the ambient sense input could be turned up for a similar effect but you risk running out of headroom; it doesn't provide the accuracy of the Threshold; and it's a poor practice if you plan to install calibrated systems like the one described next. To get more system output, amplifier sensitivity could also be increased. This practice is also discouraged because it defeats system calibration and the protection provided by the Max Gain settings.

Attack Time and Decay Time are the last settings you'll need to configure. **Attack Time** can be thought of as a sensitivity setting. It determines how quickly the *AMB-5* allows the system gain to rise from the Ambient Offset to the peak noise response. **Decay Time** is similar, but it determines how quickly the *AMB-5* allows the system gain to be reduced. To configure these settings "on the fly," start with the shortest Attack Time (0.2 seconds) and listen while the system is operating to determine whether the setting is appropriate. If the *AMB-5* responds too quickly, increase the Attack Time and perform listening tests until the response time is acceptable. Configure Decay Time in the same way: start with a short Decay Time (0.2 seconds), do listening tests, and increase the Decay Time until it satisfies your requirements.

The only other setting is the Guard Band. The **Guard Band** is simply a "scratch pad" for the Threshold, so it will usually be set to 0 (zero). The Guard Band offsets the Threshold directly, so a setting of 0 (zero) is required when you don't want to offset it.

5.1.2 Calibrated Setup

If SPL levels are defined in the sound system specification, you might find it's faster and easier to calibrate the AMB *algo* to absolute SPL rather than setting it by trial-and-error. When calibrated, the *AMB-5*'s Channel 6 bar meter (visible on the Display screen) will correspond to a real sound pressure level. For example, you could calibrate the ambient input so 0 dBu on the bar meter corresponds to 100 dB SPL. Any Threshold you set will then correspond to a real sound pressure level and Expansion Ratios can be easily translated into real increases in SPL. Calibration is recommended particularly when a single ambient mic will be used to control each *AMB-5*. This is because it's much more difficult to calibrate multiple mics on a single ambient channel. You shouldn't calibrate such a system unless you have extensive experience in sound system calibration.

For a single mic system, you'll need an SPL meter, pink noise generator and an amplified, full-spectrum loudspeaker. Once you've installed the ambient mic, pick a location that corresponds to the location of the noise source. Use the pink noise generator to drive the loudspeaker to a good reference level, such as 80 dB SPL, and use the SPL meter to take measurements from the location of the ambient mic. On the computer, bring up the bar meter display for the *AMB-5*. Adjust the hardware gain potentiometer of the ambient sense input (Input 6) on the back of the *AMB-5* until the measured 80 dB SPL corresponds to a good reference meter reading, such as –20 dBu. An input bar meter reading of 0 dB will now correspond to 100 dB SPL.

A calibrated system let's you do things that can't be done "on the fly." For example, if a local safety ordinance happened to require an emergency paging level as loud as the loudest noise source, you could calculate the settings quickly by taking a few measurements. First, use the SPL meter to measure the loudest noise source from the critical listening area (a critical measuring location may be specified in the local ordinance). Second, set up an Ambient Offset that works well when there are no trains in the station. Third, measure the pink noise output of the public address system during low noise operation from the same location in step 1.

The low noise measurement is the low end of the ambience system's dynamic range. For the high end, the Max Gain settings must be high enough to drive the system to the loudest measured noise level. You can set Max Gain easily by running pink noise through the public address system and increasing Max Gain until SPL measurements from the critical listening area are equal to the loudest measured noise level. Next, set a Threshold that's above the pink noise level you observed during low noise operation. Often, there is a range of settings that could be used as a Threshold, so you'll need to make sure the Threshold setting allows enough headroom for the ambient input to drive the system level up to Max Gain. The best way to test the headroom is to set the Expansion Ratio to 1:1 and watch to see if the system gain rises to the Max Gain levels during high noise operation. If an Expansion Ratio of 1:1 doesn't drive the system to Max Gain, the Threshold should be lowered.

When measuring from the location of the ambient mic, the subway train has a peak noise level of 105 dB SPL and the crowd has a peak noise level of 75 dB SPL. As it turns out, a good Threshold level is halfway between the high and low noise levels, or about 90 dB SPL (–10 dBu). The range from the Threshold to the peak train noise level is 15 dB SPL, which is a good range for a 1:1 Expansion Ratio. The 1:1 Expansion Ratio is used to provide an increase in sound system gain for every increase in ambient noise as specified in the local ordinance.

On the platform where people need to hear the sound system, the train's peak level is about 102 dB SPL, so the Max Gain settings are configured to provide a slightly

higher output of 105 dB SPL in the critical listening area. In the critical listening area, crowd noise peaks at 83 dB SPL. The Ambient Offset is set to –14 dBu (86 dB SPL) for low noise operation.

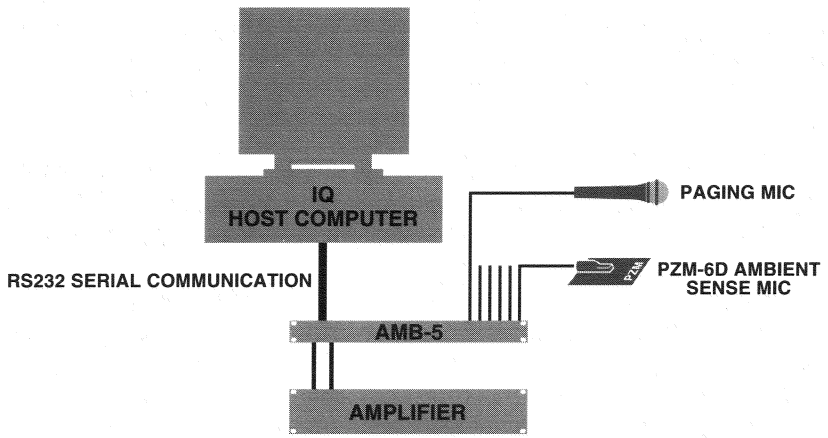
There are literally hundreds of effective ways to approach system setup. In the last example, a Threshold was chosen to accommodate a 1:1 Expansion Ratio which was part of the specification. In other circumstances, it might be better to calculate an Expansion Ratio that accommodates a particular Threshold. The Max Gain setting was set up to cut off the output at the highest required sound pressure level—a different approach might be to set Max Gain at the highest output level that's safe and try different Expansion Ratio settings.

The examples here are provided to help get you started, and to stimulate your thinking. Just remember the most important rule: The proof of the pudding is in the eating—no matter how much you think about it, no matter how good it looks on paper, you have to listen to a sound system to assure that it works well and sounds good.

5.2 Subway *IQ System* Design 2

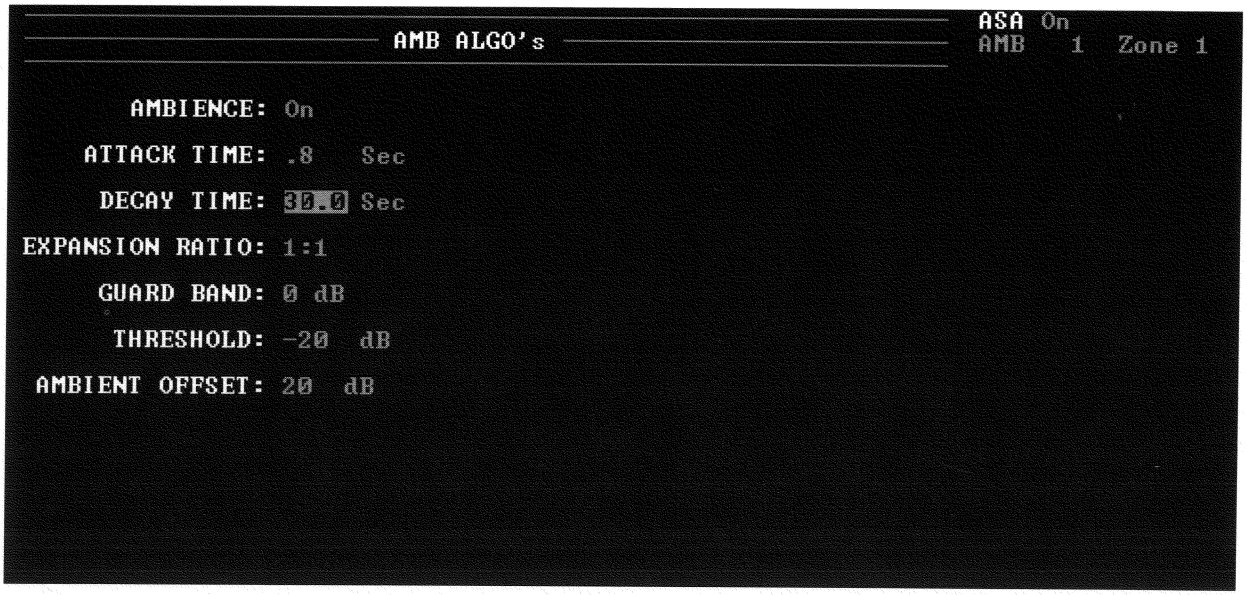
This design is for a subway station that's larger than the one in the first example. The primary system requirement is intelligibility. Because of the size of the station, it's impossible to track noise levels with a single ambient mic. When the train leaves the tunnel, the distance between the platform and the mic makes noise levels too low. More ambient mics could be added, but this would increase the expense and complexity of the system. So, a similar hardware setup is used as the first subway design.

Subway System



Fortunately, a long Decay Time can be used to accommodate this situation. Each subway train that passes through stops for approximately 20 seconds. After the initial sound of the train activates the ambience system, a long Decay Time is used to maintain higher system gain levels even after the noise falls below the Threshold. This way a single ambient mic effectively controls a preset adjustment time.

The preferred method of operating the *AMB-5* is to let it track ambient levels directly without using a preset decay time. The *AMB-5* will then respond very closely to the exact ambient levels as they occur over time. Although the single mic approach costs less than a multiple mic system, there is an associated reduction in signal control. Furthermore, larger installations usually require additional ambient microphones regardless of any tricks you might try. In this situation, a close tracking would only be possible if the subway station were smaller or if another mic were added.



Max Gain and the Ambient Offset are set first by trial-and-error. Next, the Threshold should be set. In this design, the Threshold can be set higher than in the last example because (1) the required adjustment range is smaller and (2) the Threshold's purpose is to report the presence of a train—not to track the noise level. There is no attempt to track the train's noise once it leaves the tunnel and stops at the platform.

The remaining *algos* are set by ear as well. The Attack Time should be short so that the system responds to the train noise immediately. An immediate response should be safe because a high Threshold should prevent other noises from affecting it. The Expansion Ratio will probably need to be set to 1:1. With such a high Threshold, the system will need as much gain expansion as possible. And again, if an Expansion Ratio of 1:1 doesn't provide enough system gain, it might be necessary to lower the Threshold. As mentioned earlier, the Ambience Algo Pack can only provide increases in system gain that correspond to ambience levels above the Threshold.

The last step is to set the Decay Time at the longest possible setting (30 seconds) and then fine-tune it by ear. When the train passes by the ambient mic, the ambience adjustment will be activated and the long Decay Time will allow paging to be tested and adjusted. Because the subway trains stop for about 20 seconds, the initial 30 second Decay Time actually works very well.

As you might have guessed, this system design doesn't work well with variations. For example, if a train stops for double the usual time, the system will return to normal gain levels before the train has left the station. Depending on the installation, this could present a problem. To overcome the situation, additional ambient mics could be added as described in the factory design that follows.

5.3 The Factory Sound System

System design becomes more challenging as noise sources become varied and you tackle larger acoustical spaces. A large manufacturing facility might have a dozen different noise sources and a dozen areas with different needs.

Clear specifications are crucial when designing more complex sound systems. Depending on the site, there might also be laws that regulate sound system design. If

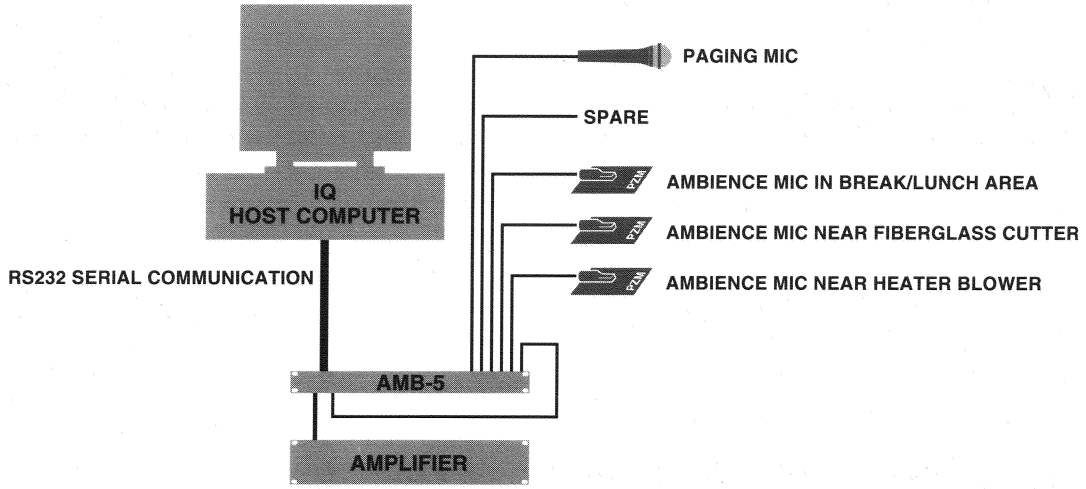
several critical areas in the factory have significantly different ambient levels, it might be necessary to use an *AMB-5* in each area. This is particularly true if the areas are separated by sound barriers, such as walls.

In this example of a factory sound system, there are no legal requirements for emergency paging. The main objective of the new sound system is to improve intelligibility. An ideal design would divide the factory space into the major noise areas and provide a dedicated ambient sense mic, amplifier channel and loudspeaker for each area. Sometimes, however, it is impossible to design the ideal system and stay within a customer's budget.

The factory is open and paging levels aren't specified for each area, so it's possible to do the job effectively by using several ambient sense mics and a single *AMB-5*. Multiple ambient mics provide several advantages over a single mic system, like the ability to "weight" each area. Also, by using multiple ambient mics, you won't need a single mic to pick up every noise source. This means you can decrease each mic's sensitivity which reduces the risk of feedback and runaway gain increases.

Like all of Crown's multiplexers, the *AMB-5* can literally function as two separate mixers. For example, inputs 1 through 3 could be mixed to output Channel 1, and Inputs 4 through 6 could be mixed to output Channel 2. With the *AMB-5*, however, the mix on Channel 2 doesn't have any algo pack control. This makes it valuable for use as a monitor mix, or a mixing channel for multiple ambient sensing mics.

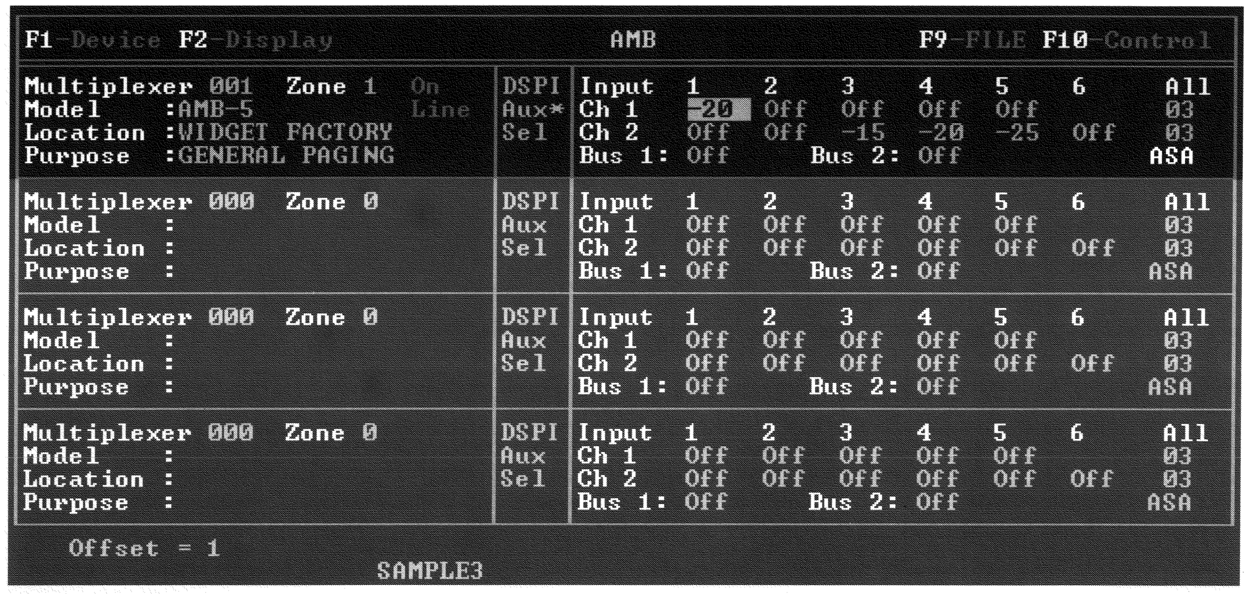
Factory Sound System

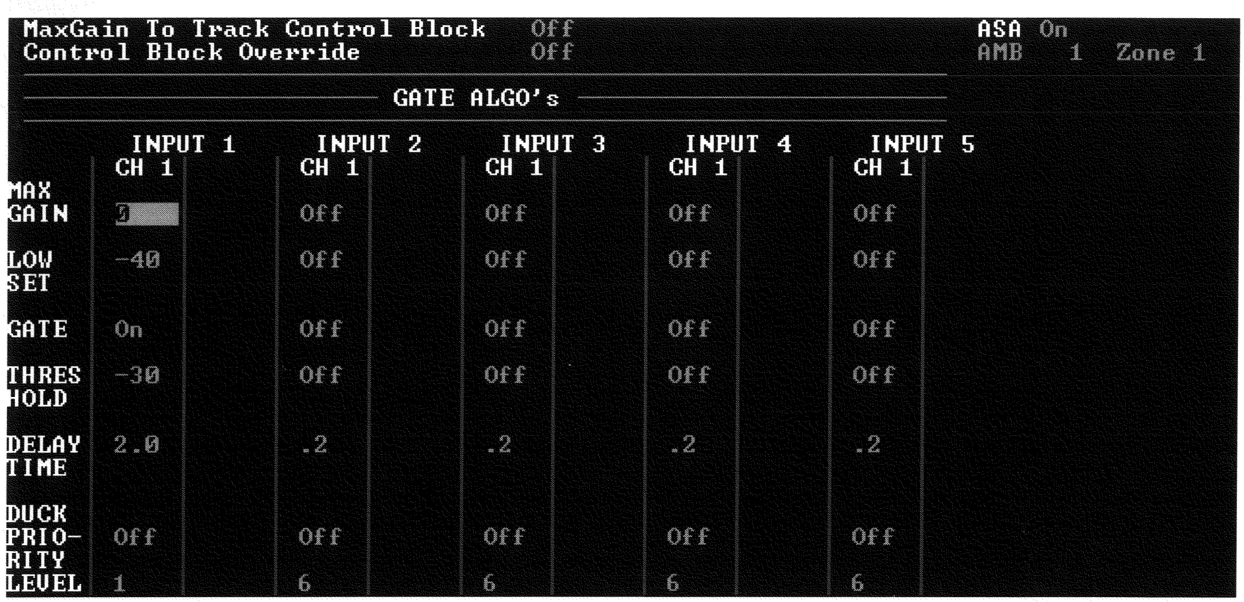


For this installation, three ambient sense mic locations have been chosen: one near the break/lunch area, one near a noisy fiberglass cutting machine, and one near a large blower for the heating system. The paging feed is connected to Input 1, and is the only signal that feeds Channel 1; the ambient sense mics are connected to Inputs 3, 4, and 5 and they feed only Channel 2.

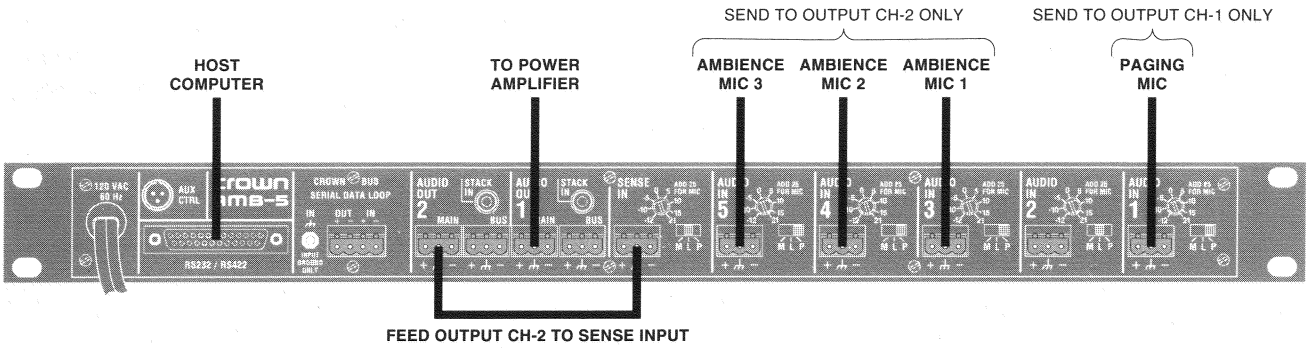
The IQ software must be used to route the inputs to the appropriate output channels. To set up this design using "two-mixers-in-one," Channel 1 must be turned off for Inputs 3, 4, and 5 because you don't want the ambient sense mics on the paging channel; and Channel 2 must be turned off for Input 1 so pages aren't interpreted as ambient noise.

Note: Before proceeding with an actual system installation like this one, be sure that you fully understand how to use the AMB-5's separate output channels as "independent mixers." If the idea of two independent mixes seems unclear, you could easily create a hardware feedback loop in the steps that follow.

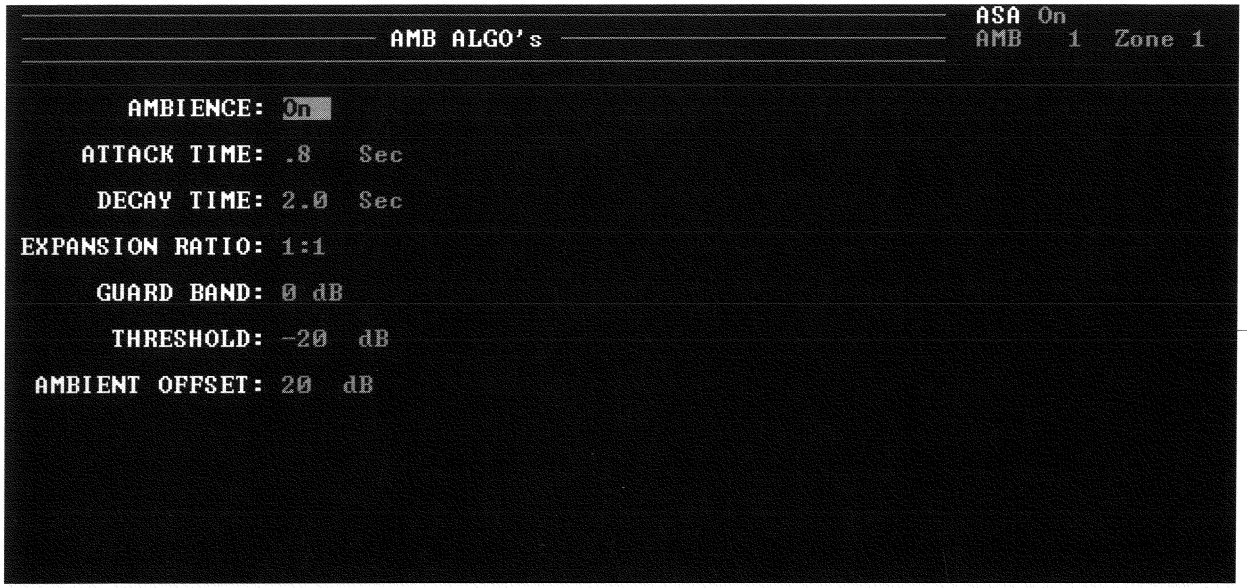




Once the ambient sense mics are wired and they are definitely <u>not</u> feeding Channel 1, you can take the output from Channel 2 and connect it to the ambient sense input (Input 6). <u>Of course, you must also make sure that the ambient sense input is not feeding Channel 2</u> (this would also cause a hardware feedback loop). Remember, the IQ software controls the routing from each input to each output channel.



When setting up the system, you should start with the paging channel (Channel 1). Turn off the AMB *algo* with the Ambience control and set the Max Gain, Gate and Compression *algos* as required for low noise levels. After the *AMB-5* is working well with the AMB *algo* turned off, the *algo* can be turned on and configured.



When mixing the ambient sense mics, first adjust each mic individually. Once each mic works well by itself, adjust all the mics together as a group. This approach normally works well because the *AMB-5*'s bar meters (on the Display screen) show peak levels, not average or RMS levels.

As in the previous examples, be sure you identify the critical listening areas. For this factory system, the two most critical areas are the lunch area and the work areas surrounding the fiberglass cutting machine. These are critical areas because they have high noise levels, but it's also important to do listening tests in other less critical areas. If you only listen to the "critical areas" with high noise levels, areas with low noise could end up with improper levels as a result.

You can use the following approaches to compensate for unbalanced listening levels: (1) if the area has a dedicated amplifier channel, adjust the amplifier output to that area, and (2) with 70-volt systems, use the loudspeaker's volume control or a different tap on its transformer. If your system doesn't offer these controls, you might want to specify more amplifiers with lower power which will provide an adjustable output channel for each area, or better yet, a 70-volt system.

5.4 Additional Ideas

Ideal system designs include an *AMB-5* with a single mic for each important area—this provides better ambience control and it's easier to set up. Also, as a result, the *AMB-5*'s Channel 2 output remains free for other purposes, such as monitoring.

Back in the subway examples, a monitoring system would probably be required. Subways always have more than one station, and most have a centralized control center for communications and management. The Channel 2 output of each *AMB-5* in the system could be used to feed a monitoring station at the central location. This way, the operation of each station's sound system can be verified from the central location at any time. And once it is decided to run monitoring cable to the central location, it might also be convenient to centralize the paging feeds. One person could make all boarding announcements from one central location where immediate information about arrivals and departures is available. The same person could also monitor the entire system to assure the system is operating properly.

Here are a few more tricks to consider. First, if you design a system to provide centralized monitoring, it's possible for you to detect system problems from a remote location using telephone data communications. With third party software and a modem installed in both your computer and the monitoring station's computer, you can call the monitoring station and look at the current system levels. You can even make adjustments to the system and upload software updates and new setup files. Of course, it might not be possible to fine-tune a system without actually being on-location, but telecomputing gives you a lot of power. You could, for example, shut down the sound system over the telephone in an emergency situation.

Another interesting design uses the AMB *algo* to control a background music source so it stays at a preset level while the paging mic responds to the ambient noise level. Set the Max Gain controls for the background music at a normal operating level and set the Ambient Offset control to 0 (zero). Now, ambience levels below the Threshold will put the background music at the Max Gain level (Ambient Offset = 0) and ambience levels over the Threshold keep the background music level at Max Gain. You could also choose to use the Duck Priority *algo* to reduce the background music during pages, but you may still need to lock in the output of the background music using Max Gain and the Ambient Offset control.

There are hundreds of *AMB-5* applications that would fill up dozens of pages. There are probably ten times as many approaches to system design and setup. This *AMB-5* applications section has been designed as an overview—it is definitely not exhaustive. You are encouraged to experiment and find system designs and setups that work best for you. So, to inspire you further, we close with a short list of applications that are ideal for the *AMB-5*:

- Subway Stations
- Factories
- Airports
- Shopping Malls
- Public Buildings
- Train Stations
- Sports Arenas
- School Lunch Rooms