## SPL Computer

A SPL (Sound Pressure Level) Computer adjusts the level of program material based upon a measurement of the ambient noise. It is typically applied where background, or foreground music must be slightly louder than a variable level crowd noise. SymNet Composer has (2) different types of SPL computer modules available; Gap Sensing and Continuous.

The term 'program material' will be used to refer to whatever audio signal is sent through the audio path of these modules. Depending on the application, this could be background/foreground music, paging signals, or a mixture of page and music, possibly with ducking.

The ambient noise measurement is taken by the SPL Computer through its "Sense In" audio input. In most applications Symetrix recommends using any omni-directional microphone for the sense input.

### Gap-Sensing SPL Computer

The Gap-sensing SPL Computer module only listens to the ambient sense microphone during program material gaps, i.e. when the program level is below a certain threshold. This prevents the module from hearing its own output which avoids run-away feedback problems.

Mono and stereo versions have the same controls. The only difference is that with stereo versions, stereo program material is adjusted. The same gain is applied equally to both channels.
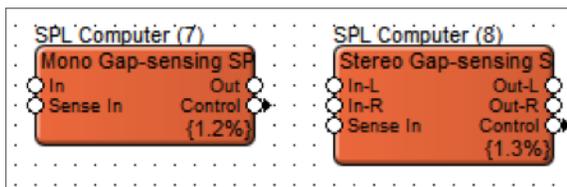


**Inputs: In** or **In-L** and **In-R**. Program material inputs. Connect the mono or stereo program material to these inputs.

**Sense In**. Sense microphone input. Connect a signal from your sensing microphone to this input.

**Outputs: Out**, or **Out-L** and **Out-R**. Program material outputs with SPL volume applied. Connect these to your main outputs.

*Note: Any equalization, dynamics processing, or level changes to the program material must be done before it is sent to the SPL computer. The SPL module should be the last thing in the signal processing chain. The gain stage from the SPL computer to the speakers must remain constant after calibration. This includes SymNet analog output block gain settings, power amplifier levels, speaker attenuators, etc... User-controlled loudspeaker attenuators should not be used when a continuous SPL computer is present.*

**Control:** This module has a control signal output. The control signal reflects the dynamic gain change being applied by the module and changes in real time. This control signal can be used to easily create linked multi-channel dynamics modules.

*Note: The control signal is scaled so that 1.0 represents the maximum gain that can be applied by the SPL computer*
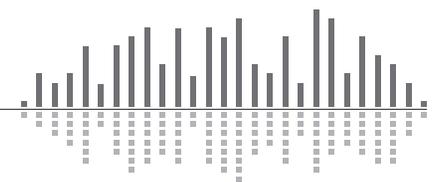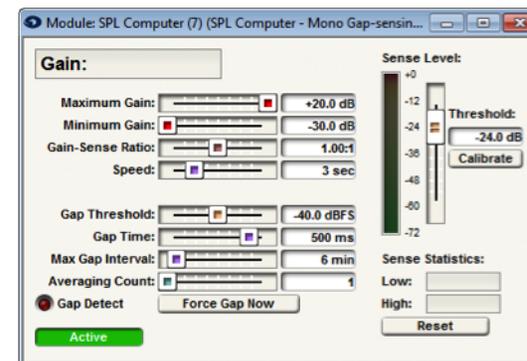
## Controls & Indicators:

**Gain**: The current gain being applied by the module (assuming it is "Active") is displayed at the top of the window.

**Maximum Gain**: dB that the SPL computer is allowed to apply to the signal. Use this control to put a limit on how loud the output can get with very loud ambient levels. Adjust using the slider or click in the text entry box to specify a numerical value.

**Note:** *The Maximum Gain cannot be set to a value less than the Minimum Gain.*

**Minimum Gain**: dB that the SPL computer is allowed to apply to the signal. Use this to put a limit on how soft the input can get with very quiet ambient levels. Adjust using the slider or click in the text entry box to specify a numerical value.

**Note:** *the Minimum Gain cannot be set to a greater than the Maximum Gain.*

**Gain-Sense Ratio**: Controls the change in gain versus the change in ambient level. Setting this to 1.0:1 means that for every 1dB increase/decrease in the ambient level, the SPLs gain is increased/decreased by 1dB. Higher values such as 2.0:1 mean that the gain is increased by more than the ambient increase, which allows out-shouting the crowd. A setting of 0.5:1 means that the gain is only changed by 0.5dB for every 1dB ambient change giving a more subtle effect.

**Speed**: Controls the rate in which the module changes the gain, specified in seconds. Longer times can cause a very gradual fade up or down in response to changing ambient levels.

**Gap Threshold**: Controls the level under which the program material must be in order for it to be considered a gap. When the program material is less than this level for at least the Gap Time, the Gap Detect LED will light and the SPL will respond to the ambient level. When the program material is above this level, the SPL will not make adjustments based on the ambient level. This level should be set so that the Gap Detect LED lights whenever the program material is soft enough that the ambient sense microphone does not pick up

a significant amount of program material, i.e. the ambient noise dominates the program pick-up. By positioning the sense microphone in order to minimize pick-up of the program material, a higher gap threshold can be set and then take sense measurements more often, e.g. in softer musical passages. If the program material has regular gaps (e.g. a paging signal or background music with clear breaks between songs) set this threshold quite low to just above the noise floor of the program material. In doing so, it will only be sensing during actual gaps in the music/paging signal.

**Gap Time**: Controls the length of time in milliseconds that the program material must be silent (below the gap threshold) to be considered a gap. This setting can be used to compensate for the reverberation time of a live room. It can prevent the algorithm from responding to the reverberation tail of a page or other program material. Settings in the 100-200ms range are good for most environments, though very live rooms may require settings of 1 second or more.

**Max Gap Interval**: If no gaps occur in the program material in this amount of time, the module forces a gap by momentarily muting the audio.

**Averaging Count**: Averages the number of designated sense readings before calculating a gain change. The Max Gap Interval is used as the sampling rate for the moving average.

**Gap Detect LED**: This LED lights when a gap is detected and the module is responding to the sense input.

**Force Gap Now** button: Pushing this button forces a gap a sense immediately by momentarily muting the audio.

**Active button**: When engaged, this button activates the module so that the gain adjustments are applied to the audio in the signal path. When inactive, SPL calculations are inhibited.
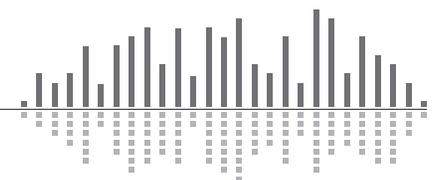
## Sense controls:

**Threshold**: The threshold sets the sense input level at which the module applies unity gain. This should be set to the average or typical ambient level of the room during calibration. The meter displays the average level of the sense input as a reference. Use the fader, or numerical entry to adjust the threshold.

**Calibrate** button: Forces the threshold to be set to the current meter reading. Ideally, calibration should be done with an average ambient level and no program material playing.

**Sense Statistics**: Shows the highest and lowest sense values logged since the last reset. Sense values are only logged during gaps.

**Reset** button: Clears statistic values. Use this feature to monitor the ambient noise levels in a room over time.

## Gap-Sensing SPL Composer Calibration Procedure:

When calibrating the Gap-Sensing SPL make sure no program material is playing. Use the Sense Level meter and Threshold fader to select and apply unity gain. Press the Calibration button to force the threshold to be set to the current meter reading. Once unity gain is set, adjust the faders for Maximum Gain (how much louder from unity gain should the SPL Computer be allowed to raise the program material gain), Minimum Gain (how much quieter should the SPL Computer be allowed to lower the program material gain), Gain-Sense Ratio, Speed, Gap Time, Max Gap Interval, and Averaging Count to achieve the desired performance of the SPL computer.

*Note: Maximum Gain will put a limit on how loud the output can get with very loud ambient levels, and Minimum Gain will put a limit on how soft the input can get with very quiet ambient levels. Maximum Gain cannot be set to a value less than the Minimum Gain, and Minimum Gain cannot be set to a greater than the Maximum Gain.*

## Continuous SPL Computer

Continuous SPL Computer listens to the ambient noise level and makes adjustments continuously. The Continuous SPL Computer can therefore be used in environments where there are no gaps in the program material. The module can also be used with program material that contains gaps, but it is recommended that the gap-sensing module be used instead. The gap sensing module is easier to calibrate, can make more appropriate adjustments according to crowd noise, and uses fewer DSP resources.

Mono and stereo versions have the same controls. The only difference is that with stereo versions, stereo program material is adjusted. The gain is applied equally to both channels.
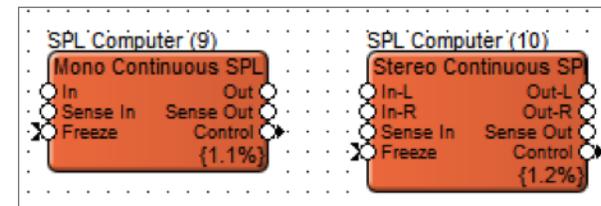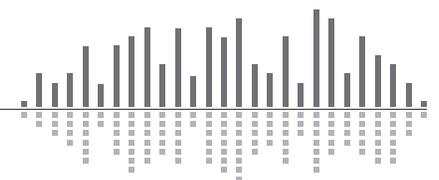


**Inputs**: **In** or **In-L** and **In-R**. Program material inputs. Connect your mono or stereo program material (foreground/background music or page) to these inputs.

**Sense In**: Sense microphone input. Connect a signal from your sensing microphone to this input.

**Freeze control signal input**: This control signal input that can be used to inhibit SPL calculations and hence freeze the SPL gain at the current level. When the Freeze input is at or above 50%, the SPL gain will be frozen (though changes to the Output Trim will still take effect). This feature may be useful for example in some paging applications to prevent gain changes during a page. If this feature is not required, this input may be left open.

**Outputs: Out** or **Out-L** and **Out-R**. Program material outputs with SPL volume applied. Connect these to your main outputs.

***Note:*** *Any equalization, dynamics processing, or level changes to the program material must be done before it is sent to the SPL computer. The SPL module should be the last thing in the signal processing chain. The gain stage from the SPL computer to the speakers must remain constant after calibration. This includes SymNet analog output block gain settings, power amplifier levels, speaker attenuators, etc... User-controlled loudspeaker attenuators should not be used when a continuous SPL computer is present.*

**Sense Out**: This is the signal that the module is using to sense ambient level changes. It is the same signal as Sense In except filtered by a voice-band (300Hz - 4kHz band-pass) filter. This signal may be used as a diagnostic to hear what the module is hearing.

**Control**: The control signal reflects the dynamic gain change being applied by the module and changes in real time. This control signal can be used to easily create linked multi-channel dynamics modules.

***Note:*** *The control signal is scaled so that 100% represents the maximum gain that can be applied by the SPL Computer.*
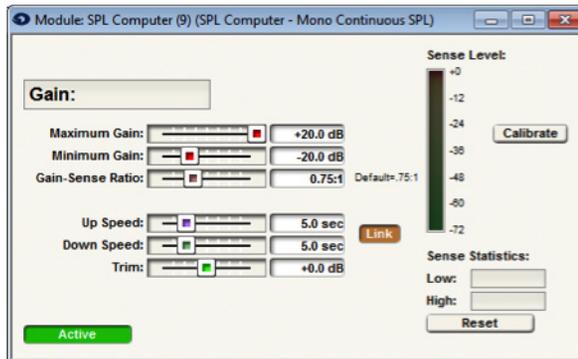


## Controls & Indicators:

**Gain**: The current gain being applied by the module (assuming it is Active) including the output trim is displayed at the top of the window.

***Note:*** *Before calibration, the display shows Gain: Not calibrated.*

**Maximum Gain**: dB that the SPL computer is allowed to apply to the signal. Use this to put a limit on how loud the output can get with very loud ambient levels. Adjust using the slider or click in the text entry box to specify a numerical value.

***Note:*** *the Maximum Gain cannot be set to a value less than the Minimum Gain.*

**Minimum Gain**: dB that the SPL computer is allowed to apply to the signal. Use this to put a limit on how soft the input can get with very quiet ambient levels. Adjust using the slider or click in the text entry box to specify a numerical value.

***Note:*** *the Minimum Gain cannot be set to a value greater than the Maximum Gain.*

**Gain-Sense Ratio**: Controls the change in gain versus the change in the ambient level. Setting this to 1.0:1 means that for every 1dB increase/decrease in the ambient noise level, the SPL's gain is increased/decreased by 1dB. Higher values such as 1.5:1 mean that the gain is increased by more than the ambient increase, which allows "out-shouting the crowd." A setting of 0.5:1 means that the gain is only changed by 0.5dB for every 1dB ambient change giving a more subtle effect. Use the lowest setting that works for your application, and *use caution with gain: sense ratios above 1:1, since these are more likely to be unstable.*

**Up Speed**: Controls the rate at which the module increases the gain, specified in seconds. Longer times can cause a very gradual fade up in response to increasing ambient levels. Technically, the time indicates how long it takes for the gain to change by 10dB, e.g. a setting of one second means a 10dB/second gain change.
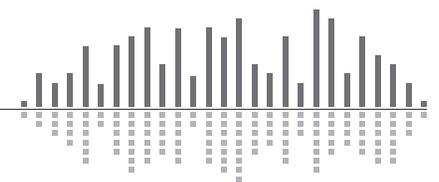
**Down Speed**: Controls the rate at which the module decreases the gain, specified in seconds. Longer times can cause a very gradual fade down in response to decreasing ambient levels. Technically, the time indicates how long it takes for the gain to change by 10dB, e.g. setting of one second means a 10dB/second gain change.

*Note: The settings of Up and Down Speed also have an effect the averaging time of the module. The lesser of the two speed settings is used to control the averaging time.*

**Link** button: When depressed, this button links the up and down speeds so that they can be moved together. This button only applies to changes made from SymNet Composer, not from external control. In many installations the up and down speeds will be the same. Separate controls are provided for situations where the noise level changes asymmetrically, e.g. a train terminal that fills slowly, but empties quickly.

**Trim**: This parameter allows for manual gain adjustments. This output trim is applied on top of the gain that the SPL algorithm dictates, i.e. the actual gain applied is the SPL calculated gain plus the output trim, limited by the maximum and minimum gain applied is the SPL calculated gain, plus the output trim, limited by the maximum and minimum gain settings. The effect of the output trim is indicated in the Gain display. Before and during calibration, this setting is ignored and treated as if it were set to zero. When the SPL calculations are frozen, changes to output trim will still take effect. If you need to give an end user some control over volume with an SPL computer module in use, this is the place to do it.

**Active button**: When engaged, this button activates the module so that the gain adjustments are applied to the audio in the signal path. When inactive, SPL calculations are inhibited, just as if the Freeze control input was on.

## Sense controls:

**Sense Level Meter**: This meter displays the current RMS level of the sense input as a reference.

**Calibrate button**: Press this button to initiate the calibration process. The SPL Calibration Wizard opens and steps you through calibration.

**Sense Statistics**: shows the highest and lowest sense values logged since the last reset. Sense values are logged continuously, and this value represents the raw sense input consisting of both ambient noise and program material pickup.

**Reset button**: Clears these values. Use this feature to monitor the ambient noise levels in a room over time.

## How the Continuous SPL Computer Works:

Since a continuous SPL computer makes adjustments while the program material is playing, it must be able to distinguish what is program material and what is ambient noise. In an ideal world, the SPL sense input would be a signal that contains nothing but ambient noise.. However, in the real world, physical constraints make this impossible. To determine how much of the signal at the sense inputs ambient noise vs. program material a calibration procedure is used. The calibration procedure makes a known change in the program level and measures the corresponding change in the sense input level. By comparing these values it determines the feedback gain, that is, the gain between the SPL outputs and the sense input. (For this reason, the gain between the SPL module outputs and the sense input should remain constant during and after calibration. Remember, a trim control on the Continuous SPL Computer allows for end user/integrator program material gain changes that will not negatively affect the calibrated operation) During normal operation, the SPL computer algorithm monitors its own output signal, and knowing the feedback gain, it knows how much of its own signal it is receiving back at the sense input. After accounting for that, the remaining signal at the sense input is the ambient noise component.

# SYMETRIX SOLUTIONS: TECH TIP
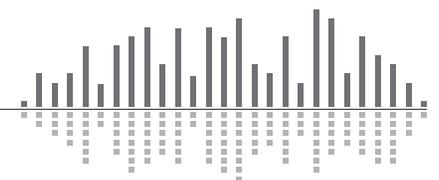
July 2014

## Sense Input Considerations:

The sense input is typically generated by one or more microphones. If multiple microphones are used, mix them together before sending to the sense input. The microphones and speakers should be configured so that the microphones hear a maximum of ambient noise and a minimum of program material. Directional microphones with the speakers in their rejection axis may be useful. While the algorithm can discriminate between noise and program material within reason, the less program material pickup there is, the more accurate and stable operation will be. In the perfect situation with 100% ambient noise and 0% program material, gain adjustments will be perfectly accurate and stable. If the program material becomes louder than the ambient noise, the module's operation is degraded. It is recommended that the program material level be equal to the ambient noise level, or at worst be no more than 6dB louder than the ambient noise level for proper operation. In an extreme case where the sense microphone picks up nearly 100% music/page (0% ambient noise), the module will not be able to extract enough information to use for gain control and calibration will fail.

## Continuous SPL Composer Calibration Procedure:

SymNet Composer steps you through calibrating the SPL Computer module. Calibration needs to be performed before the module will function. Before calibration, the module applies unity gain and the gain display shows "Gain: Not calibrated". (Gain: Not calibrated displayed on the Continuous version only.)

Calibration should be performed when the ambient noise level is at a typical volume. This noise level will become the unity gain point, i.e. the ambient noise level at calibration corresponds to unity gain by the module. Also, must have valid signals at both the "In" and "Sense In" inputs to calibrate. Make sure the microphone is working and generating a reasonable level, which can be verified using the Sense Level meter. Also make sure that there is typical program material preset as well. If program material is not available, use a pink noise generator as the program material. (In fact pink noise is a very good signal to calibrate with since it contains a broad range of frequencies and is at a constant level.) The program material should remain at a relatively constant level during the calibration procedure.
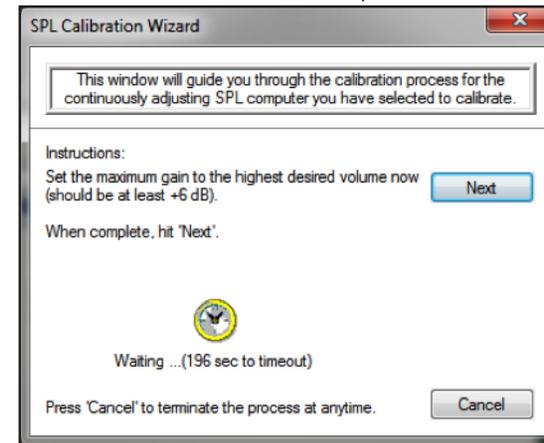
To initiate calibration, press the Calibrate button. The SPL Calibration Wizard opens, then walks through the calibration process. The first screen allows canceling out of the process if calibration was started accidentally. Hitting the Start button begins the calibration in earnest. The first thing that happens is that the gain is set to unity and measurements of the program and ambient noise levels are made. After this, adjust the Maximum Gain slider to the desired setting. You will hear the program audio increasing as you increase this setting. You should set the value to at least +6dB for the calibration procedure to work. (This maximum gain can be adjusted later if necessary.) After the maximum gain setting, hit the Next button. If calibration succeeds, a success message will be presented. If it fails, the reason for the failure will be listed. Some common examples are insufficient level at either the program or ambient inputs.
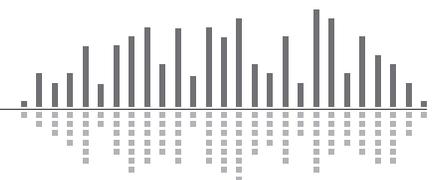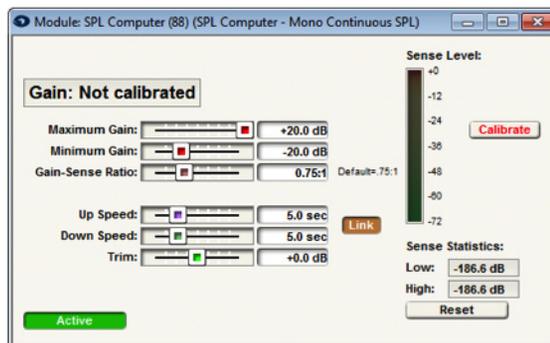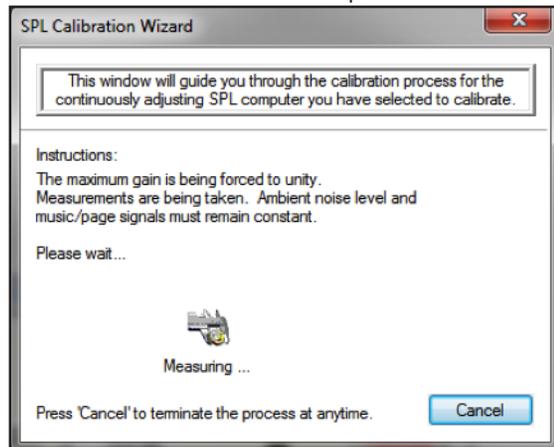
Calibration Step 1:
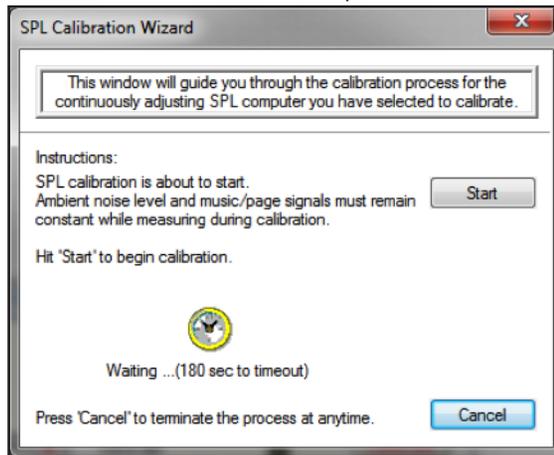


Calibration Step 2:



Calibration Step 3:



The ambient noise level should remain constant during calibration so that the change due to the increased program material can be clearly heard. If during calibration, the crowd starts yelling or someone drops a platter of dishes, re-calibrate!

Once calibration is performed, the calibration data will be retained even through re-downloading to the SymNet hardware. If the unit is due to be power cycled, be sure to check the "Last Known Operating State" bubble under Power On Control States in the Tools/Site Preferences dialogue of SymNet Designer, in order to retain the calibration settings through the power cycle.

In SymNet Composer go to Tools->Site Preferences and check the "Last Known Operating State" option for the Power On Control States section.
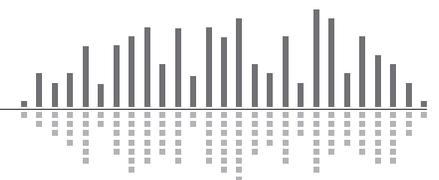
## Tips for Difficult Situations:

Great efforts have been made to ensure that the Continuous SPL Computer module does not run-away, that is hear its own signal, increase the gain, leading to hearing more of its own signal, etc. until the gain becomes stuck at maximum. However, in difficult situations where the sense input contains a very large portion of program material, some instability may result. If the gain changes unpredictably or tends to get dramatically too loud or soft, try these tips.

- Re-calibrate. If anything in the sound system has changed since calibration, this may be throwing off the algorithm. Re-calibrating with normal ambient and program material levels will often fix the problem.

- Adjust the microphone/loudspeaker placement to minimize program material pickup. Fixing things acoustically is usually the best remedy. Switching to a directional microphone or adding sound-absorbing material may help. Get a rough idea of what the sense input is hearing just by listening to the Sense Out of the module. If there is a lot of program material and very little else, that is a sign of trouble. Also, at the end of calibration, some statistics are presented that give a more exact indication of noise vs. program material pickup.

- Use a lower Gain-Sense ratio. Lower ratios mean less potential for wild gain changes. Ratios above 1:1 should especially be used with caution. The default of 0.75:1 is good for many applications.  Use the lowest ratio you can get away with.

- Use slower speeds. Using larger times for the up and down speeds causes the module to average out level changes over a longer period of time. The slower the speed, the more stable the module will be. Use the slowest speeds that can get away with. Also, if the module is tending to increase gain too much, try making the Down Time shorter than the Up time.

- Flatten the frequency response in the path from SPL output to sense input. (This is an advanced technique and usually isn't necessary.) The more closely the picked up program material appearing at the sense input matches the program material leaving the SPL module outputs, the more accurate and stable the module will be. First tune the room suing an equalizer so the main loudspeaker outputs are relatively flat. Compensate for the frequency response of the sense microphone by putting an equalizer on the sense input signal.  Send pink noise, swept sine waves, or whatever method normally used to tune a room through the SPL computer module. Then adjust the sense input equalizer so that the signal going to the sense input is as flat as possible in the region from about 250Hz-5kHz. This should ideally be done in a quiet room.

- If there is some situation where a priority page could be present through the system at a very high level, freeze the SPL gain during the page. The Freeze control signal input is provided for this purpose. A level detector or other control logic module could be used to detect when the page is happening and freeze the level of the program material.

## Continuous SPL Technical Notes:

If the calibration procedure succeeds, some statistics are presented showing parameters measured during calibration. These are intended to be diagnostics to help tech support solve problems, but a brief description is given here for the curious. The Music/page level and Noise level values tell how much of the sense signal was calculated to be program material (music/page) vs. ambient noise during the unity gain portion of calibration. Ideally, the noise should be louder than the music page, and the greater the difference the better the module will work. This number can give an indication of how well the microphones are placed to hear the ambient noise and reject the program material.

A feedback gain parameter in dB is also presented. This shows the overall "loop gain" from the SPL module output, through the loudspeakers, through the sense microphones, and back to the SPL module sense input.
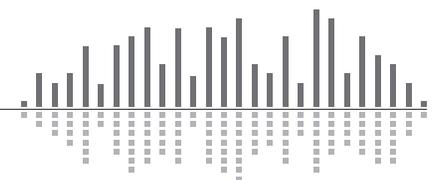
The absolute value of this number doesn't have too much significance, but if calibration is repeated this number should stay relatively constant. If it doesn't this could indicate that the module is having trouble distinguishing the program material from the ambient noise. Try calibrating with pink noise instead of lowering the program material level to see if it can achieve more consistent results.

## What You Should Not Do After Continuous SPL Calibration

Do not adjust the level of any volume controls in the sound system signal path following (downstream from) the Continuous SPL Computer module. Doing so will cause the module to yield erroneous results. This includes controls both inside the SymNet unit and those outside of it such as equalizer settings, amplifier input level controls, wall-mounted L-Pad style speaker attenuators, etc. Also, do not insert compressors into the signal path after the Continuous SPL Computer module. If adjustment is needed for amplifier input levels or equalizer settings, re-calibrate the module afterward so the module can adjust to the new levels. If trim is needed for the overall system level up or down after the calibration, the preferred method is to adjust the module's Output Trim control, through adjusting the signal level upstream from preferred method is to adjust the module's Output Trim control, though adjusting the signal level upstream from the Continuous SPL Computer module works as well. Do not adjust the gain downstream from the SPL module! Obviously, user-controlled speaker attenuators should not be used with a Continuous SPL computer. Similarly, equalizers after the SPL module should remain fixed after calibration. If adjustable equalization is needed, e.g. treble/bass controls, place the filter modules prior to (upstream from) the SPL computer so it knows about them.

## Things to Consider for Both Gap-sensing and Continuous SPL Computer Operation:

### Location of Microphone:

Much more important than the type of microphone is its location. The sensing microphone needs to "hear' the ambient sound within the controlled space. It is vital that the microphone is placed where it primarily picks up a majority of noise rather than the paging or music that is going through the system. Do not locate the sensing microphone near a localized noise source that is not typical of the ambient noise level of the controlled zone.

For example, the noise from a large machine of some sort, a kids play area, or a video game, may cause the module to think that the zone is noisier than it really is. The best sense mic placement ensures that the majority of the signal picked up by the sensing microphone is ambient noise. As mentioned above, some programs material pickup is acceptable, but the less there is, the better the module will work.

### Signal Path Considerations:

The best place to put the SPL Computer module in the SymNet module signal chain is that the very end, right before the output. If this is not possible for some reason, make sure the signal path after the SPL computer is fixed after calibration. For example, if a graphic EQ is used to flatten room response, complete the EQ tuning procedure before calibrating the SPL computer module and then leave the EQ controls fixed afterward. Speaker protection limiters are OK as well, as long as they are set up so they are rarely reducing the gain (i.e. they are not consistently being driven to the point of limiting).

## When to choose Gap-sensing vs Continuous SPL Computers:

Gap-sensing SPL Computer modules only listens to the ambient sense microphone during program material gaps and is best suited for a restaurant or bar type environments. The SPL Computer module will use the ambient noise level of the room between program material and then increase or decrease the overall program level. Since an averaging count can be set to determine the gain change applied by the SPL Computer, the program material gain changes can be set to change with the trending ambient noise SPL levels rather than adjusting to a specific ambient noise measurement. For instance, the averaging count could be set to 5 to insure a honking horn next to a restaurant in NYC does not cause the SPL computer to raise the program material gain to it upper limit just because the Gap-sensing SPL Computer took a sample of ambient noise at the time of the honking car.

Continuous SPL Computer modules make adjustments while the program material is playing and is best suited for a stadium type environment where ambient noise level is constantly changing while the program material is playing.