# Noise Sensing Using a Variation of the nLMS Adaptive Filter with Auto Calibration

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This White Paper discusses a system that will compensate for the noise level in a room by measuring the program and noise level sensed through an ambient microphone. The system then changes the program level in proportion to the noise level that was sensed through the microphone. The technique that is presented here uses a combination of analog to digital conversion (ADC), adaptive digital filtering running on a digital signal processor (DSP), and digital to analog conversion (DAC). The adaptive filter employed is a variation on the Normalized Least Mean Squares (nLMS) method. This approach effectively "nulls out" any music that was sensed at the ambient microphone after which the only thing that remains is the noise. A Root Mean Square (RMS) measure of this noise level provides the ability to adjust the program level accordingly.

## 0 Introduction

Virtually anyone who has ever listened to music in an automobile has realized this fundamental fact: while driving, the music level must be louder than while the car is parked. The reason for this is because while driving the noise level (wind, road, etc.) is louder than while parked. This requires that the listener constantly adjust the music level to compensate for the varying noise levels. This is not only a problem for automobiles but virtually all sound systems where background noise is varying substantially. For example: in a factory setting, the music would be set to one level while the machinery is running and another while it is not running. For most systems, this requires that someone always adjust the level in proportion to the noise. The question naturally arises: "why can't this be done automatically?" Mackie Designs has invested a considerable amount of time in research and development to find an answer to this very question. In the process, we have developed a sophisticated DSP noise sensing algorithm that will perform this task precisely. Mackie's SP-DSP1™ is an automatic level controller that maximizes intelligibility by changing gain in proportion to environmental noise level changes [6]. Basically, this system senses the level of the ambient noise of a room and adjusts the system gain accordingly. To work properly, the controller must "null out" any effect that the program material (music) has on the noise being received by the ambient microphone. The method we have employed to differentiate the noise from the program material is what makes our algorithm unique (patent pending). One innovative feature of our algorithm is that it adapts over time to the varying room acoustics (i.e. people, drapes, sliding doors, etc.) to provide the best possible music rejection. This significantly reduces the possibility of "runaway" gain as exhibited in existing hardware-based implementations [10]. To accomplish this, we have utilized a combination of digital hardware (SP-DSP1™) running a complex software algorithm [8]. Figure 0 shows the hardware block diagram of the noise sensor. The software is actually twofold: an embedded software algorithm plus application software (SP-Control™ for the Palm™) to allow for ease of user control. Additionally, the SP-DSP1™ algorithm allows for a high level of automation, which in-turn, makes this system extremely easy to setup and use. Unlike some of the earlier attempts at "noise sensing", the SP-Control™ software requires no complex procedures during setup and calibration. The user simply places his speaker(s) as needed and positions the ambient microphone so that it is listening to the primary noise source. Then the appropriate gain structure is setup as well as a few room-specific user parameters. Finally, while playing music, an *Auto Calibration* is initiated. It's fast and simple!

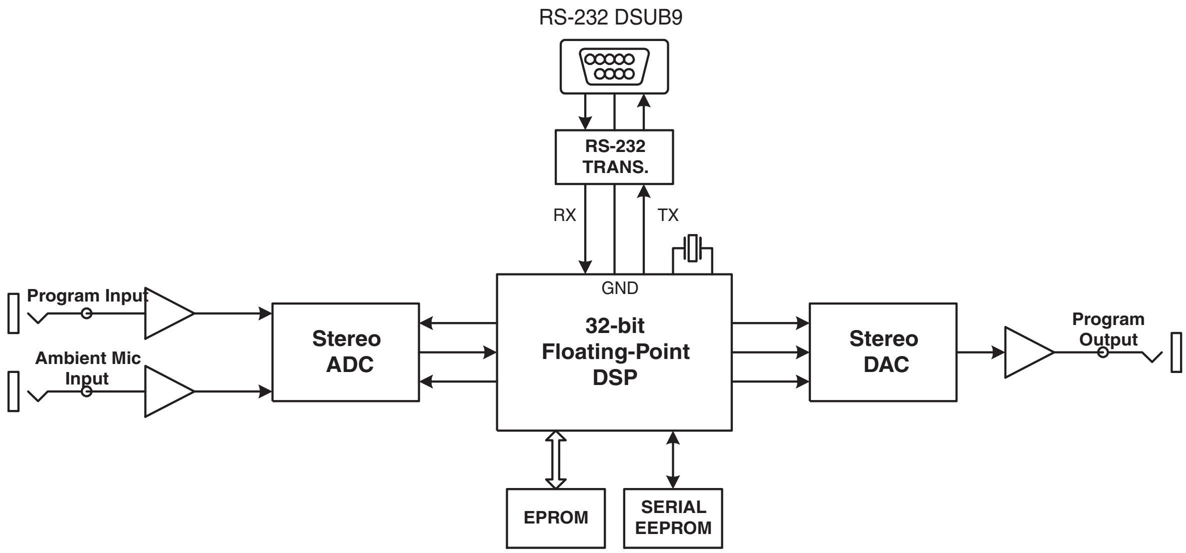


**Figure 0: Hardware Block Diagram**

## 1 Algorithm Overview

This section summarizes the Ambient Noise Sensing Algorithm (see Figure 1) implemented in the SP-DSP1™. The algorithm is used to increase the sensitivity to ambient noise in the room by rejecting the source signal (music) that is broadcast into the room. This allows the SP-DSP1™ to control the volume of the music based on the room ambient noise. The better the rejection of the music signal the more sensitive the gain control without runaway gain problems.

The algorithm adapts to the room characteristics by comparing the room response to the source signal. It computes its own approximation of the room response in order to cancel the music signal from the signal picked up by a room microphone [1-5,7]. Room size is the most important factor determining the effectiveness of the algorithm. Small rooms (office size) can achieve as much as 40dB music rejection, while larger rooms (nightclubs or restaurants) might only achieve 10-20dB. Even 10dB rejection allows noticeably better sensitivity to noise.

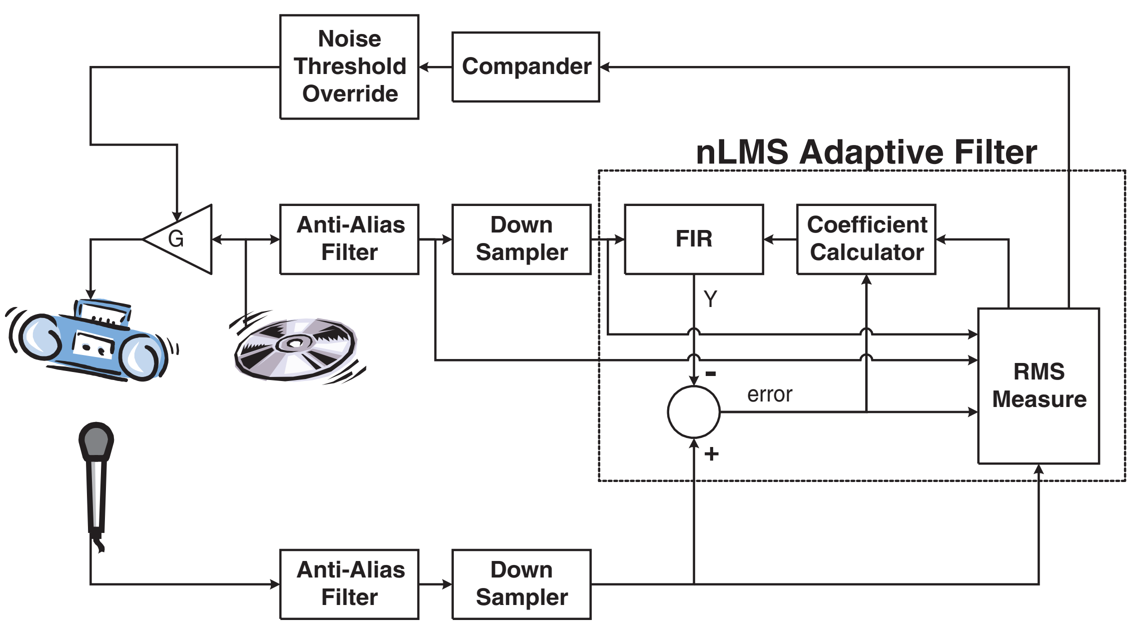


**Figure 1: Algorithm Block Diagram**

## 1.1 Algorithm Block Diagram

The ambient noise sensor algorithm is a variation of a Normalized LMS Adaptive Filter [2-4, 7, 11-12]. The coefficient calculator in the system uses the error signal, some Root Mean Square (RMS) signal levels and energies, and the current coefficients to compute a new set of Finite Impulse Response (FIR) coefficients that approach the Room Transfer Function (RTF) [5]. Down sampling has been used to optimize the adaptation characteristics of the algorithm for the largest room possible using minimum MIPS and memory [8, 9]. The compander controls the gain of the music signal based on the ambient noise detected and the user input parameters.

## 1.2 Normalized LMS Adaptive Filter

The Normalized Least Mean Square (nLMS) Adaptive Filter portion of the Ambient Noise Sensor consists of the FIR filter (Equation 1), Coefficient Calculator (Equation 2), Summation (producing the error signal), and RMS measurements, and can be considered equivalent to the MATLAB[1] simulation shown in Diagram 1 [1, 9, 10]. In this simulation the 'Voice' signal is the ambient noise signal, the delays represent the RTF, and the nLMS filter adapts to produce the 'Out' signal which is an approximation of the music combined with the RTF. When the output signal from the nLMS is subtracted from the input signal (Voice + RTF (Music)) the 'err' or error signal approximates the original 'Voice' or noise signal. Look at the gains and delays shown in the blocks of Diagram 1 (simulating the direct sound plus one reflection). The amplitude versus time plot was generated with the coefficients from the 'Taps' output of nLMS filter. As you can see from this simple example, this filter is the RTF. Obviously, in the real world, the exact RTF is not obtainable. However, the better the FIR approximates the RTF, the closer the error signal is to the actual noise in the room [2-4, 11-12]. The nLMS Adaptive Filter is constantly adapting to the room characteristics. This provides optimum performance when the room acoustics change. Room acoustics can change significantly due to the arrangement of furnishings, opening or closing drapes, the number of people in the room changing, or more dramatic changes such as moveable walls or room dividers.

The effectiveness of the nLMS is determined mostly by the number of taps in the FIR (internal to the nLMS adaptive Filter), the adaptation algorithm, and the density and duration of the RTF. In the SP-DSP1™ the processing capability and memory allocation has limited the FIR length to about 1500 taps. As a result, the signals are down sampled to allow for good cancellation even in reasonably large rooms. Since the speed of sound is approximately 1 ft/ms, we designed the system to provide adequate rejection for music paths (direct sound plus reflections) up to about 300 feet.

*[1] MATLAB is the trademark of a comercial matrix-based processing language.*

A standard Transversal n-point FIR filter was used [1, 5, 8]. Input buffering is n points long plus a little extra buffer space. The Output, y, forms another buffer:

$$y(n) = \sum_{i=0}^{N-1} w_i(n)*u(n-i) \tag{1}$$

where

**N** = number of FIR taps or the order of the filter
**n** = time index
**y** = FIR output
**w** = adaptation coefficients
**u** = FIR input

Calculation of new coefficients (coefficient calculator) is performed if the following two conditions are met:

- If the normalization energy is greater than zero (if negative or 0 it is a debug error condition).
- If the Error Threshold signal is above the threshold computed as a portion of the normalization energy of the input signal.

Mathematical calculations for coefficient updating are according to a slightly modified nLMS formula [2-4, 7, 11-12]:

$$W_i(n+1) = W_i(n) + \frac{e(n) * beta * u(n-i)}{epsilon * E(n)} \tag{2}$$

where

**n** = time index
**i** = 0..n-1
**W** = FIR filter coefficients
**e** = error
**u** = FIR input data
**E** = Normalization Energy
**beta** = adaptation rate (function)
**epsilon** = stability coefficient

Our slightly different use of **beta** (not shown) plus the use of an Error Threshold are unique adaptations of the standard nLMS algorithm. The new **beta** function allows faster adaptation when the error, **e**, gets smaller. The normal use of **beta** as a scaling value (shown above) requires a small value to guarantee stability of the algorithm when the error is large,

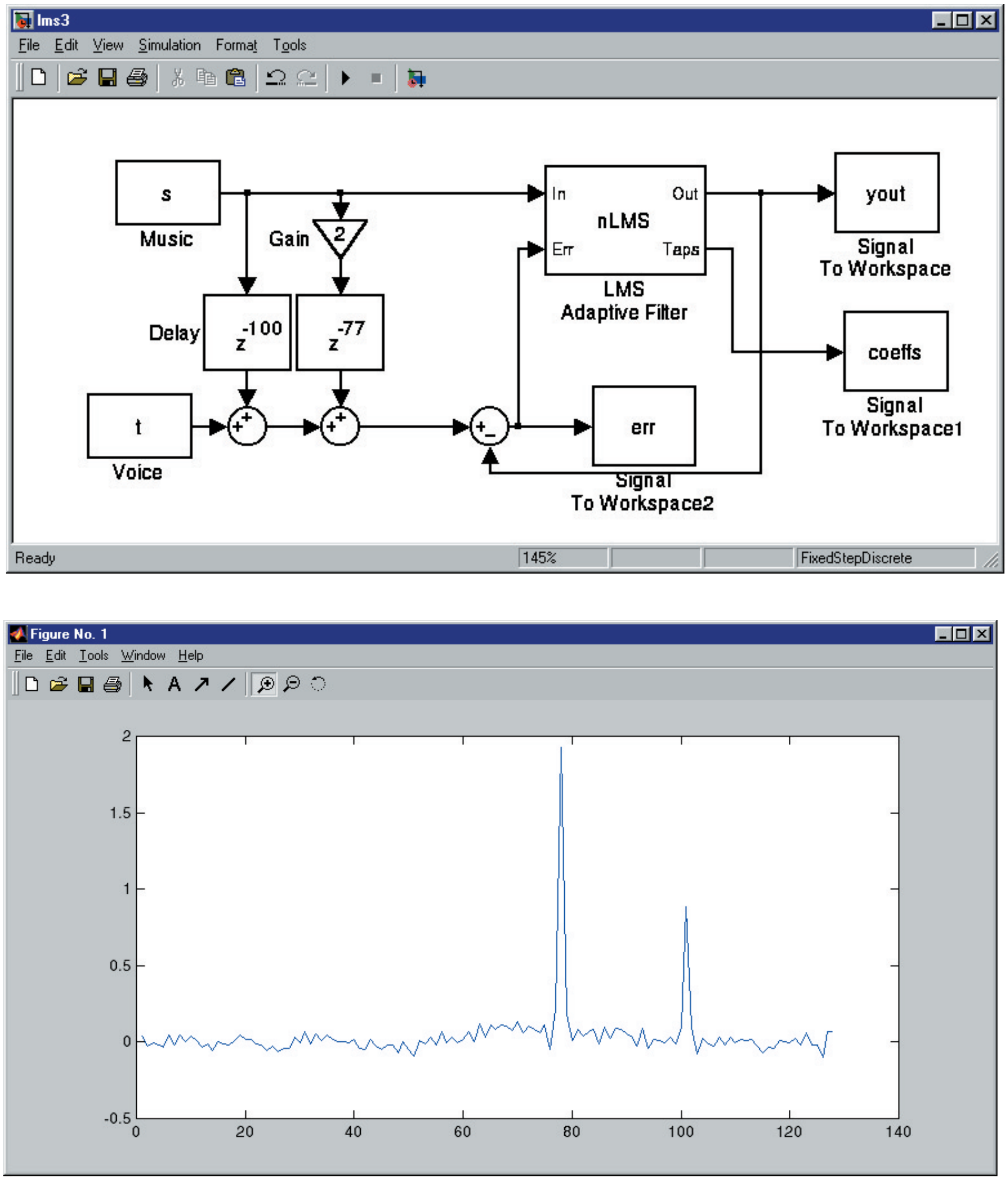


**Diagram 1: Matlab simulation Block Diagram and Results**

which significantly reduced the adaptation amount when the error was small, extending the adaptation time to well over 30 minutes. Our **beta** function (patent pending) gives stability to the algorithm when the error is large, but effectively shortens the adaptation time dramatically when the error becomes small. This allows the algorithm to adapt to a room in only a few minutes.

For small rooms a significant portion of the music energy due to the paths from speaker to microphone (including reflections) are removed by the nLMS algorithm, as there would have to be many reflections before the sound could have traveled this far. Each reflection reduces the energy of the music signal, and the amount of reduction of the energy is related to the shape and material of the reflecting surface. The more 'live' the room, the longer and more dense the RTF is, the less the energy of the music signal that can be removed by the nLMS filter [6]. Even without a lot of reflections a very large room may have poor cancellation due to the small number of reflections before the music signal has traveled the maximum 'distance' of the FIR.

## 1.3 RMS Measurements

The nLMS algorithm requires normalized energies for calculating the FIR coefficients. A real RMS algorithm is used to measure the signal levels [2, 4, 11-12]. This is accomplished by buffering the squares of each sample of the input signal (Square), averaging the sum of these squares every sample (Mean), and taking the square root of the mean (Root). For every input sample an RMS value is produced, which is itself buffered and filtered by a 256-point window function, to smooth the response of the detector by a fixed known delay and amplitude.

The main benefit of filtered RMS measurement is to reduce the impact of low frequency room energy variations caused by the music broadcast into the room, which can give variations in the room energy readings by 2 or 3 dB. This amount of variation could cause the compander to constantly turn the volume of the music up and down by a similar amount, which is quite noticeable and undesirable.

Several RMS detectors are required in the system. The nLMS Coefficient Calculator requires the energy of the FIR output, error and music signals. The Compander requires the level of the music and microphone signals.

## 1.4 Down Sampling

To extend the effective FIR length for larger rooms the LMS filter is down sampled from the converter sample rate [8,9]. In this implementation, 44100 Hz (Fs) is the converter sample rate for the music and microphone signals. Down sampling also helps reduce the memory and MIPS requirement. This also has the effect of comparing energies of signals filtered at half of the down sampled Nyquist rate. As long as the noise signal has a proportional amount of its energy in this spectrum, the ambient noise sensor will approximate the level of the noise signal. As we have found, this is a valid assumption.

## 1.5 Compander

User parameters control the operation of the compander. *Minimum Gain*, *Gain Range*, *Noise Threshold*, *Noise Range*, *Attack/Release Times*, and RMS measurement parameters determine how much gain (or attenuation) is applied to the music signal.

The compander *Attack* and *Release* parameters control the rate at which the gain is turned up or down. These are important to control the gain during loud sudden events such as door slams, yells, or dropped dish trays, which would normally cause the music level to be turned up very loud. A slow attack rate (more than 10 seconds) with a faster release rate will reduce the level of gain applied to the music. This allows the compander to track the ambient room noise while 'rejecting' these singular events if desired.

## 1.6 Auto-Calibration

The biggest single problem with controlling the music gain based on the room noise is runaway gain. Runaway gain occurs when the compander turns up the music volume which is measured as 'room noise', which tells the compander to further increase the music volume. The compander must have the appropriate information to prevent this cycle from occuring.

A calibrated algorithm for computing the approximate location at which runaway gain might occur is used as an override to limit the sensitivity to the room noise. Although the algorithm constantly adapts to the room acoustics it takes a significant amount of time for the algorithm to get to its best approximation of the RTF. Until this point is reached it is not possible to determine the override conditions required to prevent runaway gain. This condition would occur every time the unit was reset (power on/off) were it not for the *Auto Calibration* process, and the internal non-volatile storage of the calibration parameters.

The *Auto Calibration* uses a real music signal with fixed gain to adapt and monitor the ambient room noise. As long as the algorithm is making progress towards the RTF it will continue to adapt and monitor the noise level. During calibration it is best to have a minimum amount of room noise so that the algorithm can determine the progress towards the RTF. Once the progress towards the RTF has slowed significantly the current adaptation coefficients are stored and the Noise Threshold Override is computed. Every time the unit is reset the calibration coefficients and Noise Threshold Override are restored. This allows the compander to prevent runaway gain on reset and during normal operation. The algorithm continues to adapt at all times to keep up with room acoustic changes, and as long as the room acoustics do not change drastically the gain should be prevented from running away.

# 2 Hardware Setup

The SP-DSP1™ was designed to be an expansion card that is added to our new "SP" or Sound Palette® series mixer/amplifiers (SP2400/1200). These amplifiers provide one or two zones of 200 watts per channel in a two-rack-space package. Since the SP-DSP1™ operates **only** on a mono program signal, one card is required per channel. Once the card is installed, the user needs to connect an ambient microphone to the rear panel of SP2400/1200. Note that the connection requires a 3.08mm 3-pin male Phoenix connector (i.e. Euroblock).

### 2.1 Ambient Microphone

Mackie Designs recommends the Mackie Industrial MT3100® omnidirectional Electret microphone. This is an excellent multi-standard phantom condenser microphone perfectly suited for noise sensing applications. However, for many applications, any inexpensive low-impedance, omnidirectional microphone will suffice. An installer may find that a directional microphone is better suited for installations in smaller or noisier rooms. The microphone should be placed where it can "hear" the ambient noise within the room. This does not imply that the SP-DSP1™ can not differentiate between the noise and the program material. However, to get the best possible performance and rejection, the ambient microphone should be placed where it is listening to the "intended" ambient noise source. Any mechanical feedback (direct vibrations) from speaker to microphone should be avoided as well as placing the microphone away from the speakers to provide the best noise-to-signal ratio.

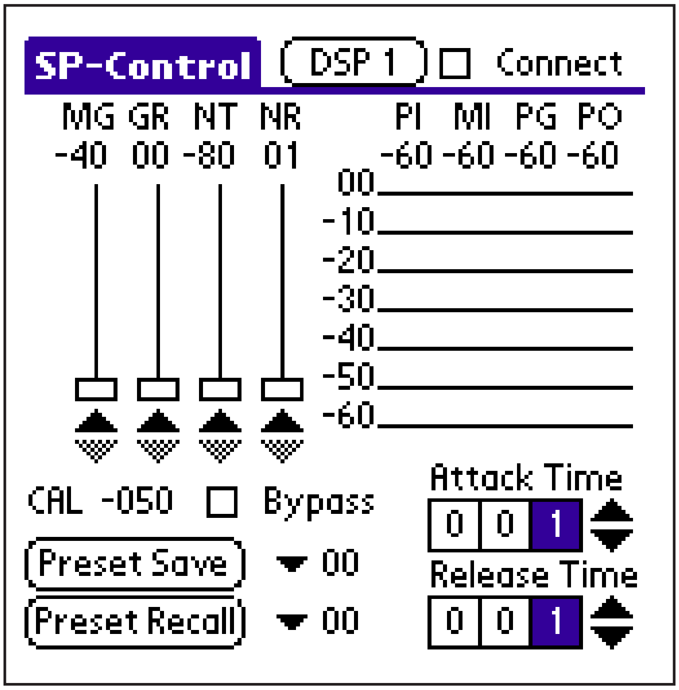


**Figure 2 SP-Control™ Main Screen**

## 3 Palm™ Control

Mackie Designs has designed the SP-DSP1™ to run under the Palm™ OS or any compatible palmtop device. The SP-Control™ Palm™ application (on 3 1/2 inch disk) is included with the SP-DSP1™ hardware card and can be downloaded from our WEB site at *www.mackieindustrial.com*. All necessary cabling is provided to install the card into the SP2400/1200. Additionally, a 3-inch null-modem adapter cable is provided to connect a standard Palm™ Cradle or the HotSync® Cable to the 9-pin female D-Sub on the front of the SP2400/1200. After installing the application to your device, to run the SP-Control™ software, simply select the "SP-Control" from the available applications. You should see the main SP-Control™ window (see Figure 2). The main display has six user parameters and four bar graphs indicating the relevant levels. Because of limited screen space, we used two-letter symbols designating the parameters and levels. Table 1 summarizes the six parameter ranges and their default settings after a *Factory Restore*.

### 3.1 User Parameters (Sliders)

As the parameter *Minimum Gain* implies, this is the lowest level to which the system can attenuate. With this slider default at –40dB, the system can attenuate the program input by as much as 40dB. Please note that the SP-DSP1™ can only attenuate program material that is present on the input. That is, this system (see Figure 0) will never allow gain from input to output (≤ 0dB). By setting the *Minimum Gain* and the second parameter, *Gain Range*, you are actually establishing the program operating window within which the levels for the program material must remain. For example, if the user wanted his music levels to operate in a range ±10dB around -15dB down from the input level, he would set *MG*=-25dB and *GR*=20.

| | *"MG"*= *Min. Gain* (dB) | *"GR"*= *Gain Range* (dB) | *"NT"*= *Noise Threshold* (dB) | *"NR"*= *Noise Range* (dB) | *Attack/ Release Time* (sec.) |
|---|---|---|---|---|---|
| Minimum: | -40 | 0 | -80 | 1 | 1 |
| Maximum: | 0 | 40 | 0 | 60 | 300 |
| Default: | -40 | 40 | -40 | 40 | 1 |

**Table 1: SP-Control™ User Parameter Ranges and Defaults**

The second parameter, *Gain Range*, also sets the maximum gain attainable from the *Minimum Gain*. That is:

**Prog. Operating Window = *Min. Gain* + *Gain Range* = Max. Gain** (3)

The *Gain Range* slider "re-scales" so that system gain (input to output) is never greater than 0dB. So, if *MG*=-10dB, then *GR*≤10dB. In most cases, the user can ignore the *Gain Range* parameter. He would simply set the desired *Minimum Gain* of the music (either by listening or from calculation) and leave the *Gain Range* at the default "re-scaled" setting.

The third parameter, *Noise Threshold*, sets the noise "trigger" of the algorithm. Until the noise level exceeds this threshold, the throughput program level is unaffected. This slider actually sets the "noise" sensitivity of the algorithm. *NT* is relative to the unity point which is –10dBFs (from full scale). Therefore, if *NT*=-80dB, this would indicate a noise threshold level of –90dBFs (where Fs = 1V RMS). In simplistic terms, the lower the *Noise Threshold* (more negative), the more sensitive program level changes are to noise level variations. The *Noise Threshold* parameter in conjunction with the fourth parameter, *Noise Range*, actually sets the operating window of the noise source. This noise "window" sets the level and range that the noise must be within to effect the program level:

**Noise Window = Noise Threshold + Noise Range** (4)

The *Noise Range* setting gives the user control over how much noise level change causes the program level to span it's *Gain Range*. By setting a small *Noise Range* (say 20dB) with regard to the *Gain Range* (say 30dB), you are allowing a 20dB noise level change to cause the program output level to span 30dB. This is assuming that the noise level has crossed the *Noise Threshold*. In other words, these settings would represent a 2:3 noise level to program level change (see Chart 1). For every 2dB change in noise level, the output level changes by 3dB. On the other hand, *NR*=40 and *GR*=40 (default settings)

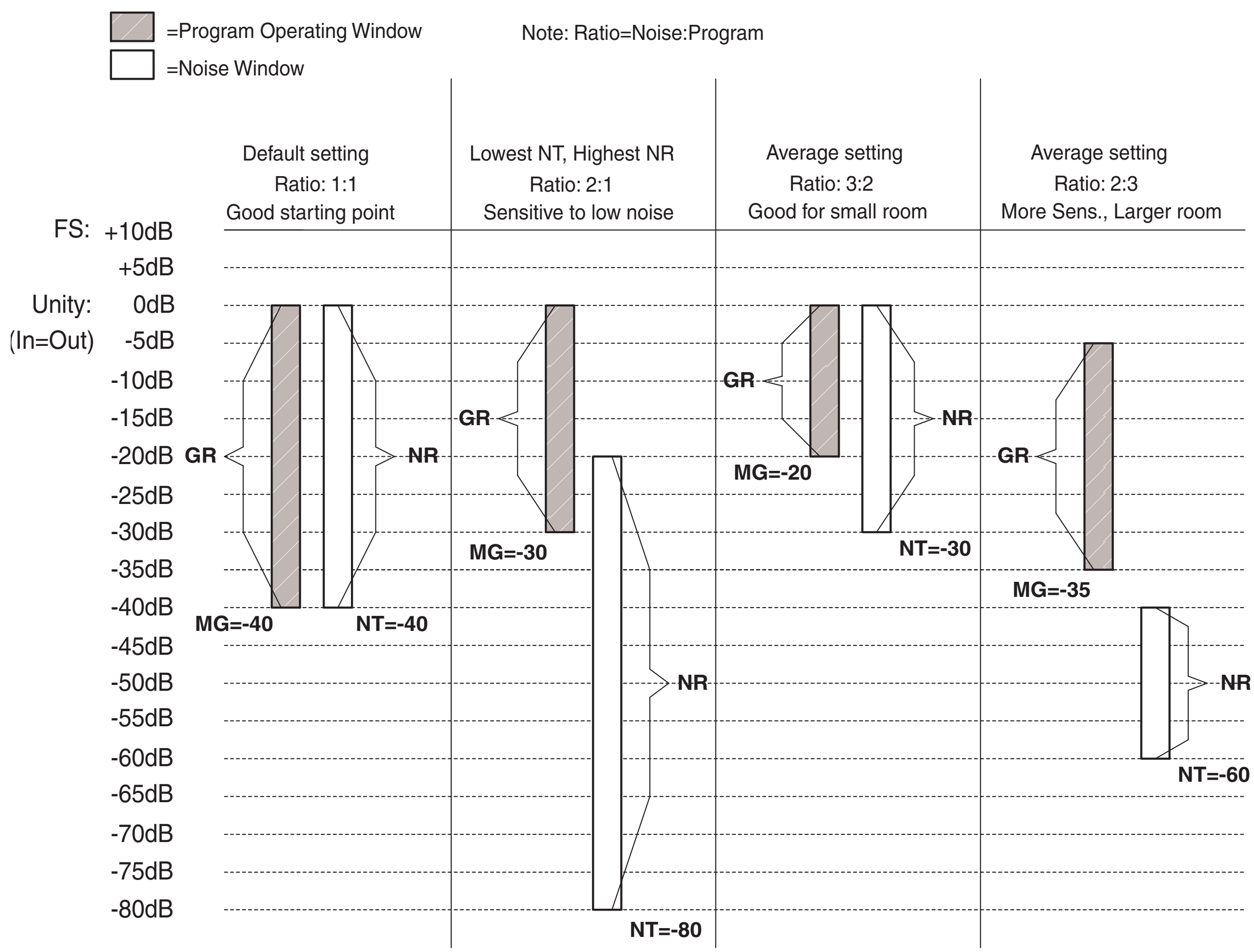


**Chart 1: Noise Window vs. Program Operating Window Examples**

would represent a 1:1 noise level versus program level change. You can start to see the flexibility of this system (see Chart 1).

By having control over the program operating window (*MG* and *GR*) and the noise window (*NT* and *NR*), the user can actually move the noise window with respect to the program operating window and visa versa. In this way, the system can be set to be "sensitive" to small noise changes or "insensitive" to large noise changes and everything in between.

## 3.2 Attack and Release Parameters

The *Attack* parameter defines the time, in seconds, it takes for the *Program Gain* to increase by 40dB. Likewise, the *Release* parameter defines the time it takes for the *Program Gain* to decrease by 40dB. The combination of these two parameters gives the user the ability to "average" the noise to provide a relatively constant *Program Gain* over time. Otherwise, with short attack and release times (i.e. the default settings), the user will hear every program level shift as it relates to a specific noise level shift. To keep impulsive noise bursts from affecting the program level, a good setting would be an *Attack*=20 seconds and *Release*=20 seconds. This would avoid the impulsive change but would still allow a fairly "responsive" program level change as it relates to a noise level change (~40 seconds). In a plant setting where there are numerous pages and "break time" buzzers, setting the *Attack* and *Release Time* on the order of minutes, will eliminate program level changes on all but the average long-term noise level. There may be a situation where you want a long *Attack* and a short *Release Time* or visa versa. For example: in a plant with background music, equipment may be turned on sequentially (relatively slowly) in the morning but in the evening all are switched off together. A long *Attack* but short *Release Time* might be advantageous by allowing the music level to decrease instantly when the noise level fades.

## 3.3 Bar Graphs/Metering

The main screen of the SP-Control™ Palm™ application has four meters that allow the user to monitor levels during setup and normal operation (see Figure 2). "*PI*" is the *Program Input* meter and it indicates the input level into the DSP (range from 0 to 60dB). During setup before *Auto Calibration*, the user needs to ensure that the input program source has sufficient signal level by monitoring this meter. The input level should be between 0 and –10dB during the loudest portions of the program material. "*MI*" is the *Microphone Input* meter. This meter indicates the ambient microphone input level in a range from 0 to 60dB. Again, before an *Auto Calibration* is executed, ensure that the microphone level is within the range from 0 to –10dB for the loudest noise plus program expected. On the rear panel of the SP2400/1200 amplifiers, there is an "ambient mic" trim potentiometer. Adjust this trim potentiometer (0-55dB) to get the proper levels as indicated by the "*MI*" meter. Note that the SP-DSP1™ does not have a mic preamp. Therefore, it is essential that the ambient microphone input receives a line-level signal (1 V RMS Full-scale). The third meter from the left as shown on the SP-Control™ screen is the "*PG*" meter which is the *Program Gain*. This meter represents the gain (0 to 60dB) that algorithm applies to the *Program Input*. The last meter is the "*PO*" which is the *Program Output* meter. This is the level at the output of the SP-DSP1™ (0 to 60dB). The following condition will always be true:

***Prog. Output* level = *Prog. Input* level + *Prog. Gain* (in dB)** (5)

A good way to set the proper "*MI*" level is to select the *Bypass* button on the main screen. Then, while playing typical program material, adjust the microphone preamp trim potentiometer on the rear panel of the SP2400/1200 until the "*MI*" meter is within 0 to –10dB. This allows us to adjust the level while the microphone is "hearing" both the program material plus the noise so that we do not inadvertently clip the SP-DSP1™ microphone input during calibration. Clipping the microphone or program input (ADC) will produce erratic controller behavior and should be avoided.

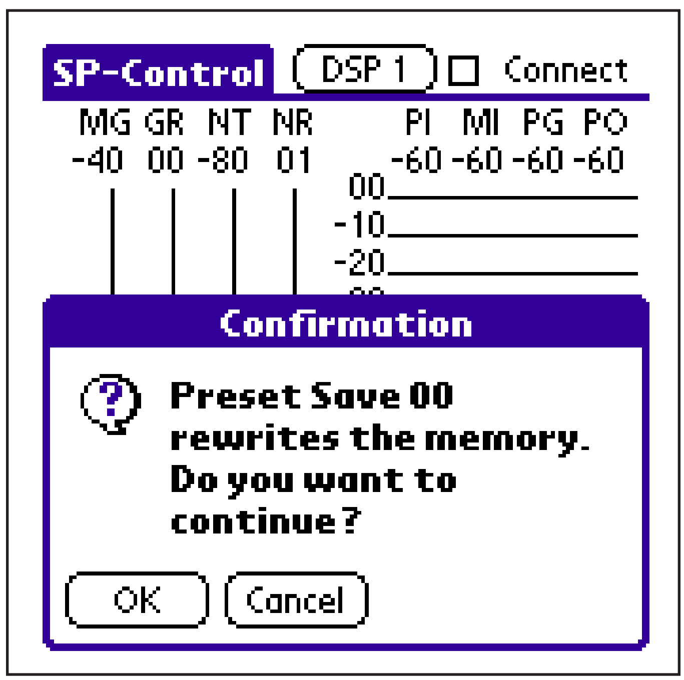


**Figure 3: SP-Control™ Save Confirmation**

### 3.4 Additional Main Screen Options

The main screen of the SP-Control™ also provides the ability to save and recall 10 presets. These presets are stored in the EEPROM onboard the SP-DSP1™ (see Figure 3). A preset consists of the following parameters: *Minimum Gain*, *Gain Range*, *Noise Threshold*, *Noise Range*, *Attack* and *Release Times.* There are two Global parameters that are recalled at power up and displayed when *Connect* is selected on the main screen. These include the *Bypass* state and the *CAL* or calibration value. Additionally, the preset that was saved or recalled when the SP-Control™ was last connected will be stored in the EEPROM and recalled at power up. The *CAL* value is set during *Auto Calibration* and will be discussed in the next section. The *Bypass* toggle, as its name implies, allows the user to bypass program input to program output and effectively disables the algorithm. This is a useful feature during setup to ensure the proper levels are obtained. Finally, the *DSP1/2* toggle allows the user to change between communicating with DSP1 or DSP2. As you are already aware, the SP2400/1200 can support two SP-DSP1™ cards; one for each zone. Therefore, if your system has two controllers, this toggle allows you to control each card separately and independently.

### 3.5 Auto Calibration

The Drop Menu of the SP-Control™ has several functions shown in Figure 4. The *Auto Calibration* provides the user with the ability to automatically adapt his speaker-microphone placement to the room acoustics (patent pending). This function eliminates the tedious calibration procedures associated with other competitor's products. Additionally, the possibility of "runaway" system gain is greatly reduced because the algorithm (a modified nLMS adaptive filter) is constantly making "running" changes to obtain the best noise-to-signal ratio. To maximize the results, an *Auto Calibration* should be performed while the room noise is at a minimum. However, since the algorithm adapts over time, this is not a critical requirement. Once the speaker(s) and ambient microphone are in their fixed locations and the microphone gain and input levels have been adjusted (per Section 3.3 Bar Graphs/Metering), simply select *Auto Calibration* while playing standard program material. Prior to performing the *Auto Calibration*, please ensure that the front panel level control on the SP2400/1200 is set to the maximum sound level that you would ever expect the system to deliver. Recall that the noise sensor only attenuates the signal from input to output. During the calibration, a count-down timer is initiated at 90 seconds and a progress meter is displayed. If the algorithm finds that adaptation improvements are possible, the timer is reset to 90 seconds. A typical calibration period is 2-3 minutes. It is advised to monitor the main four meter levels during calibration to ensure that all levels are in their target range as suggested in Section 3.3 Bar Graphs/Metering. If you find that the microphone and input levels are too low or too high, you may want to *Abort* the calibration by selecting the appropriate button and then readjust the levels accordingly. There is also an *End & Save* button if the user finds that calibration is taking too long or music breaks are causing count-down timer resets. Typically, music breaks between songs are no problem. However, if the break is too long, the calibration may be ineffective resulting in numerous timer resets. If this becomes a problem, simply changing to different program material or using music with shorter pauses between songs should correct the situation. Once the calibration is complete, the *CAL* value will update. At default, this value is +10 (range: +20 to –80dBr). This number is a measure of how much of the program material is getting rejected in the environment that the *Auto Calibration* was performed. The closer the number is to +10, the less rejection. A *CAL* = -15dBr would be an excellent rejection and would be expected in a small room. However, in larger rooms and auditoriums, the rejection will be less (closer to 0dBr). Remember that this number is only an estimate of rejection that is used by the algorithm to set the Noise Threshold Override.

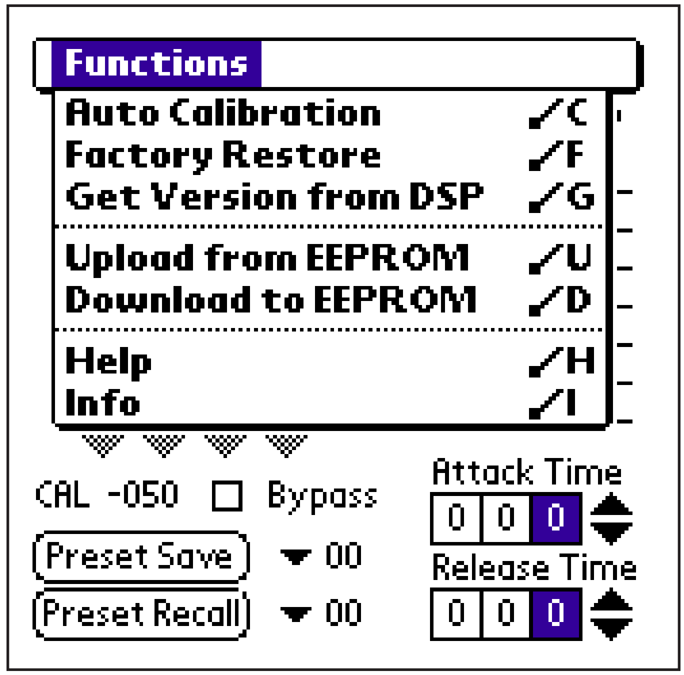


**Figure 4: SP-Control™ Function**

### 3.6 Factory Restore

If you find that the EEPROM is corrupted or has not been initialized from the factory, you can perform a *Factory Restore* upon installing the SP-DSP1™ and running the SP-Control™ application. A new SP-DSP© card should have all parameters of all presets set to default values (see Table 1).

### 3.7 Upload/Download From/To EEPROM

Occasionally, it is necessary to backup SP-DSP1™ presets and parameters or to transfer settings from one DSP to another. To copy the contents of the EEPROM, simply *Connect* to the DSP card and then select *Upload From EEPROM.* The EEPROM contents are stored in your Palm™. If you would like to copy contents of the Palm™ into another SP-DSP1™ card, select the *Download To EEPROM.* Be advised that this will completely erase the existing contents of the EEPROM and replace with what was uploaded to the Palm™. Be careful to do an *Upload From EEPROM* **before** the *Download To EEPROM* to ensure that valid data exists in the Palm™.

## 4 Tips for Setting User Parameters

To clarify, setting the user parameters (*MG, GR, NT, NR, Attack Time and Release Time*) can be done either before or after the *Auto Calibration*. These parameters have no bearing on the calibration and can be changed at any point. In fact, a *Preset Save* and *Recall* only affect these parameters.

The *Auto Calibration* only concerns itself with the Calibration or *CAL* value and the adaptation coefficients. The user parameters can be set up for runaway gain, but the user would notice this immediately as the system would “runaway” to the maximum gain. During calibration, the user parameters are temporarily overridden to perform the calibration at a fixed gain (easier and better results). However, the fixed gain used during the calibration process is the Maximum Gain (*Minimum Gain + Gain Range*). It is probably advisable for the user to set up these two parameters so that he isn’t “blown away” by the calibration sequence if his Maximum Gain is louder than he intends. The Maximum Gain is used as the calibration value because the algorithm adapts more quickly when the music is loud. The calibration itself doesn’t limit the parameters the user can adjust in any way, it just provides a ‘hidden override’ when he has his *Noise Threshold* set too sensitive, which would otherwise cause runaway. You’ll notice that once your unit has been calibrated you can set the *Noise Threshold* as low as you want and you will still seem to get the same sensitivity. This is because it is being limited by the Noise Threshold Override (see Figure 1).

As you can see, one of the first questions that the end-user needs to answer is “How much overall system gain or attenuation do you want the noise sensor to provide?” By answering this question you have determined the setting of the *Gain Range* (*GR*). A typical good starting point for the Gai*n Range* is about 20dB. If you find the system does not attenuate enough or attenuates too much, adjust accordingly. By answering this first question, you are likely to determine the *Minimum Gain* (*MG*) as well. Since the average user would want his system to attenuate from the unity point (0dB), which is the highest sound level he would ever want, for this example the *Minimum Gain* would be set to –20dB. As mentioned in Section 3.1 User Parameters (Sliders), by setting the *Minimum Gain* and the second parameter, *Gain Range*, you are actually establishing the program operating window within which the levels for the program material must remain. As in the previous example, if the user wanted his music levels to operate in a range ±10dB around -10dB down from the input level (unity point), he would set *MG*=-20dB and *GR*=20.

Setting the *Noise Threshold* (*NT*) can only be done by determining the noise level at which you want the music to start to increase in level. This noise threshold point is best found through trial and error. You must be careful not to set the *NT* too low (i.e. more negative), because the noise sensor will begin “gaining up” the program material with very little noise. However, setting the *NT* too high will prevent the system from making any program gain increases until there is a relatively high noise level. There is a balance and each room will require a different *NT* setting. Keep in mind that setting this level in an empty room may require some “tweaks” when the room is full with people. For example, if you set *NT*=-40dB in an empty restaurant, you may find that the system starts gaining up too fast and is too loud for the occasion. In this case, you can reduce *NT* (by bringing the level closer to 0dB) by 5 to 10dB. This would give you *NT*=-30 to -35dB and the system would not start “gaining up” until the noise was 10dB “hotter” than it was previously.

*Noise Range* (*NR*) actually sets the amount of noise that makes the program gain go from the minimum setting to the maximum setting. That is, if *NR*=40dB and *GR*=20dB, then for every 2dB change in the Noise, the Program level is only changed by 1dB. A good starting point is to set *NR* equal to the *GR* setting. This would give a 1:1 noise to program level change and would suffice for the majority of installations. Remember, The Noise Threshold Override will protect the system from runaway gain **even** if the *NT* and *NR* are set incorrectly

## 5 What to Avoid

Once an *Auto Calibration* has been performed, a user should never move or change the speaker-microphone placements. In fact, any changes (i.e. adding more speakers, moving equipment, significantly changing the room layout, etc.) should be avoided. If these changes are required, simply recalibrate by initiating another *Auto Calibration*. You may find that in environments that change daily (new equipment added, equipment moves, etc.) a periodic recalibration would be beneficial.

Any gain adjustments made to the microphone preamp should be done prior to *Auto Calibration*. Further-more, **all** level changes should be made prior to the DSP card. If the system gain is changed in any way post DSP card, the noise sensor perceives this as an acoustic noise disturbance. This is why the level controls on the SP2400/1200 are before the DSP. Again, any level changes made after the DSP card (i.e. power amplifier, speakers, between the preamp and power amplifier, etc.) will require a recalibration.

## 6 HyperTerminal Control of SP-DSP1™

If the user does not own a Palm™ or compatible device, he can use HyperTerminal available on any PC running Windows OS. HyperTerminal can control all the parameters previously mentioned. The null-modem adapter is not necessary as the 9-pin female D-Sub on the front of the SP2400/1200 will connect directly to a PC COMM Port. Mackie Designs will provide a one-page protocol at the customer's request.

## 7 Patent Protection

The basic principles of noise sensing presented in this paper are the subject of patent applications.

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# THE AUTHORS

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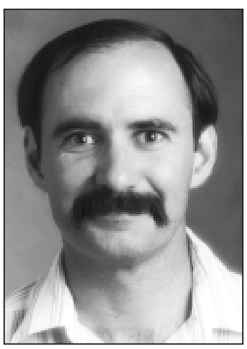

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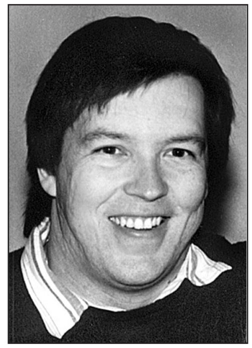

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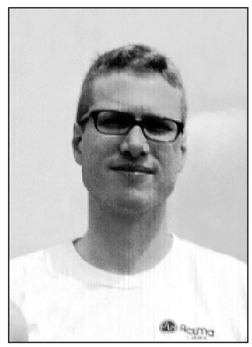

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