# Best Practices Guide

# Audio Gain Structure for Pro AV Systems


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## Who Should Read This?

In short, you should. AV technicians need to know gain structure so that they can set up systems in a better, faster, and more efficient manner. AV engineers need to know gain structure so that they can design systems with proper interfacing components, allowing the system to perform at its best. Senior AV engineers should read this as a potential resource to educate their technical staff. Salespeople should read this to obtain a cursory understanding of gain structure, and how it affects the bill of materials.

## What is Gain Structure?

Gain structure can be thought of as the sequence of level settings applied to audio signals as they pass through various devices in an audio or AV system. When proper gain structure is applied, the system as a whole performs its best, with speech, music, and other content sounding clear, dynamic, and free of distortion. Many audio devices, such as audio digital signal processors, preamplifiers, mixers, AV switchers, amplifiers, and some source devices include one or more user-accessible gain controls. Proper audio system gain structure is the result of careful selection of equipment and properly setting audio levels through each gain control within each device, allowing for optimal signal-to-noise ratio and headroom.

## Understanding Gain Structure

To help get a better sense of gain structure in an audio system, let's use plumbing as an analogy. The water that we drink originates from a water tower or reservoir, and then travels through a maze of pipes and valves before it comes flowing out of our faucets. If some or all of the valves along the way are not opened enough, we get a dribble of water from our faucet, as illustrated in Figure 1a. We are not getting enough water and the pressure is so low that the water is picking up impurities from the pipes.

If all of the valves along the way are opened too much, we get plenty of water pressure, but we run the risk of bursting any of the pipes along the way. See Figure 1b. In this case, we are getting too much.

With a mixture of valves opened too much and valves opened not enough, it is possible to get sufficient water pressure at the faucet, but the water will have impurities and we run the risk of bursting a pipe. See Figure 1c. By setting all valves to the proper pressure, we get sufficient water pressure at the faucet, and enough pressure in the pipes to ensure no impurities and the pipes won't burst, as shown in Figure 1d. This is the ideal scenario for this plumbing system.

Figure 1

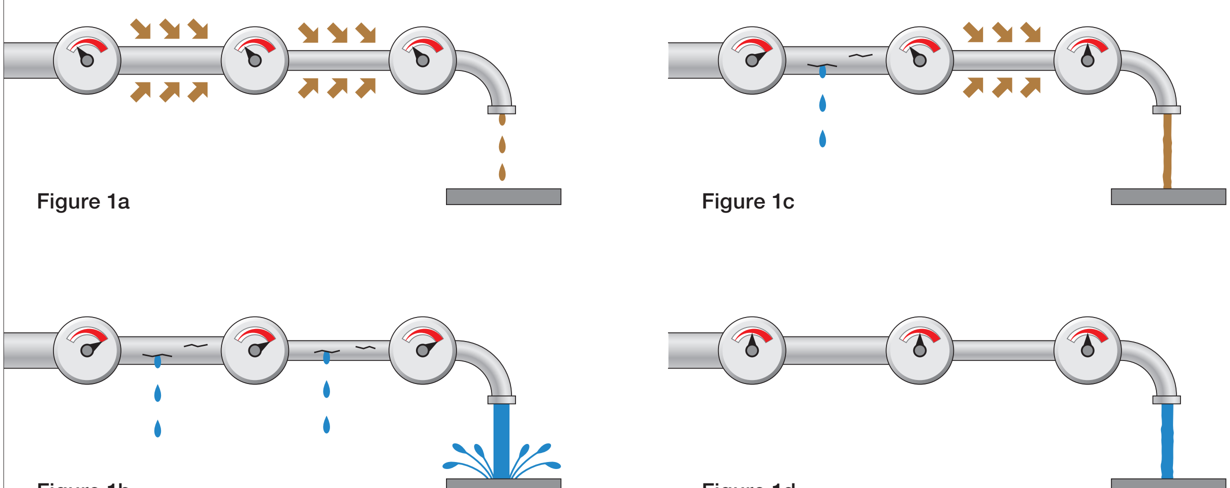


Figure 1a

Figure 1c

Figure 1b

Figure 1d

A sequence of valves in a plumbing system is an effective analogy for visualizing audio gain structure.

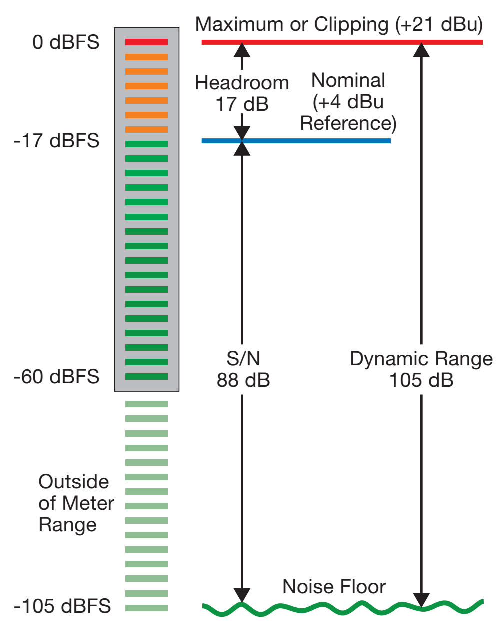
**How Does Plumbing Relate to Audio Gain Structure?**

Consider the water impurities as an equivalent to circuit noise – or in an audio system, the noise floor, which can be heard as hiss. Sufficient water pressure ensures that the quantity of impurities at our faucet is inconsequential. Similarly, having sufficient audio levels through each AV device ensures that the noise floor will not be noticeable at the loudspeakers. Consider the bursting pipes as equivalent to clipping or distortion. Too much water pressure, even in only one pipe, can put the entire plumbing system at risk of a burst pipe. With audio, signal clipping or distortion in one component will affect the entire signal path. For example, if there is clipping at a microphone input, the rest of the system will pass that distortion through with great clarity and accuracy. What you hear will be objectionable, and your customers won't be pleased.

## Why is Proper Gain Structure Needed?

Proper gain structure makes the difference between good and bad sound. Maintaining a low noise floor and preventing clipping distortion are the two most notable reasons for proper gain structure. See Figure 2.

Figure 2




Audio signal levels should be sufficiently high to ensure a very low noise floor, but also allow for ample headroom to avoid signal peaks from being clipped. Specifications and meter range are shown for an Extron DMP 128 audio DSP. Targeting a nominal or average signal level of around +4 dBu, which correlates with -17 dBFS in the DMP 128, allows for about 17 dB of headroom – more than sufficient for pro AV applications.

**What Can Be Expected With Proper Gain Structure?**

More benefits than one would expect. There is actually a return on investment when audio systems are set up properly:

- **Clear, undistorted audio** – Getting the best possible sound reproduction from the system. Voices are intelligible and the music sounds good.
- **No hiss** – How many times have you stood near a loudspeaker and heard a hiss? This is a good indication of an improperly set up gain structure.
- **A consistent mix** – Everyone will be heard equally in a conferencing application, with the far end hearing the same thing as the near end, and the recording being a good representation of both. In following proper gain structure, a well-balanced mix becomes inherent to the system.

- **Optimum acoustic echo cancellation** – Echo cancellation has come a long way in the past 10 years, but it is not foolproof. By providing an adequate signal and mix to the echo cancellation algorithm, it is better able to do its job.
- **Commissioning time is reduced** – Following a systematic procedure for gain structure minimizes troubleshooting efforts, saving money for your customer.
- **Service calls are expedited** – When an organization employs a logical and consistent approach to gain structure, they establish a common baseline for all technicians to follow. This unified approach aids the service technicians in resolving an issue in a quick and efficient manner.

**What Can Happen With Improper Gain Structure?**
Many more problems than one would expect. Unfortunately, audio systems are frequently performing below their potential because of improper or arbitrary gain settings:

- **Clipping** – Apart from the intended effects of turning up guitar amps to "eleven," distortion is not pleasing to the ear. Slight distortion can be perceived as a tonal quality issue. Many have tried to equalize it out, not realizing that it is distortion. Speech intelligibility can also be affected by distortion.
- **Excessive noise** – As discussed above, poor gain structure can increase the noise in a system to an audible degree.
- **Bad mix** – Errors in gain structure at the inputs to a DSP can make mixing and routing to various destinations much more difficult. While they can be "fixed in the mix," technicians may find setup and programming to be more complex. A technician may be able to keep track of any deficiencies and work around them. But passing important system details onto others, or remembering them three months later can be rather difficult.
- **Improper automixer operation** – An automixer manages multiple microphone signals to ensure low background noise, clear and intelligible speech, and maximum system gain before feedback. However, if the microphone signal level coming into an automixer is too low, the gate may cut off soft voices intermittently, making them sound "chopped." If the microphone level coming into the automixer is too high, the gate may not attenuate background noises such as participants shuffling their papers. Also, the automixer may be less effective in managing microphone signals when participants are conversing back and forth.
- **Difficult troubleshooting** – When service technicians are called to site, they are often given the unenviable task of finding an issue in a system they are unfamiliar with. When an AV company uses a consistent approach to audio gain structure, the service technician needs less time to became familiar with the system and can jump right to finding the issue at hand.
- **Unhappy customers** – Inconsistent audio within and between systems can quickly turn a satisfied customer into an unhappy customer.

## What Can Cause Bad Gain Structure?

There are many reasons why AV systems end up improperly calibrated for audio system gain. Audio system setup and optimization, which includes gain structure, equalization, and other procedures, requires adequate time and resources for signal measurements, system programming, and evaluation of audio performance. Ideally, this would occur after all AV equipment has been installed and connected, acoustic treatments hung on the walls, and all furniture in place.

Given all the usual procedures, tasks, and training necessary to complete the AV project on schedule, there is often very limited time available for audio system setup. This can be complicated further by construction delays pushing into fixed, immovable completion dates. AV can often be the victim of the "big squeeze" at the end of new construction projects.

With the commoditization of small and medium AV systems, aggressive AV integration budgets sometimes do not provide for adequate audio setup and optimization. This is an unfortunate but very real problem in the AV industry. It is often the case that insufficient hours devoted to proper audio setup during installation later cut into the service budget.

Common practices that lead to improper gain structure include the following:

- **Improper input gain.** This is probably the most prevalent error. Once an input gain has been set incorrectly, it usually forces more errors all the way through the system to try to compensate.
- **Amplifier input gains set wide open with DSP output gains at less than 30%.**

- **Combining consumer gear with professional gear in the same system.** Consumer gear, with a line level of -10 dBV, is approximately 12 dB lower in reference level than professional gear, with a line level of +4 dBu. Proper gain levels are essential in mixed consumer / professional systems.
- **Not adhering to unity gain throughout the system.**
- **Setting gains to counteract wiring errors, rather than checking for and fixing them first.** For a balanced audio cable, if the + or - wire is not connected, the circuit will pass audio but with a 6 dB loss. Increasing the gain of that circuit "fixes" the immediate problem, but what happens when the wiring connection is fixed later? The audio signal is now 6 dB too high, which could be enough to cause clipping distortion.

Even with sufficient time and budget available, an additional challenge remains – the need to ensure all technical staff is sufficiently trained and up to date. Professional AV firms must not only embrace audio theory, but video, control, and networking as well. Adequately training and educating technical staff requires considerable time and resources.

Many excellent resources are available that detail the theory and concepts in setting up audio systems. A list of references is provided in the Appendix. It is from this body of knowledge that we have assembled a subset of practical procedures that technicians can use without prerequisite training, and apply them to commercial and pro AV applications.

## How Do I Get Proper Gain Structure?

In a word, discipline. There certainly is much more to it, but discipline is priority number one. There is always pressure to get the systems running and playing music as soon as possible. This pressure comes from fellow technicians, other trades, anxious customers, and from within ourselves. Succumbing to this temptation can lead to a disappointing first impression.

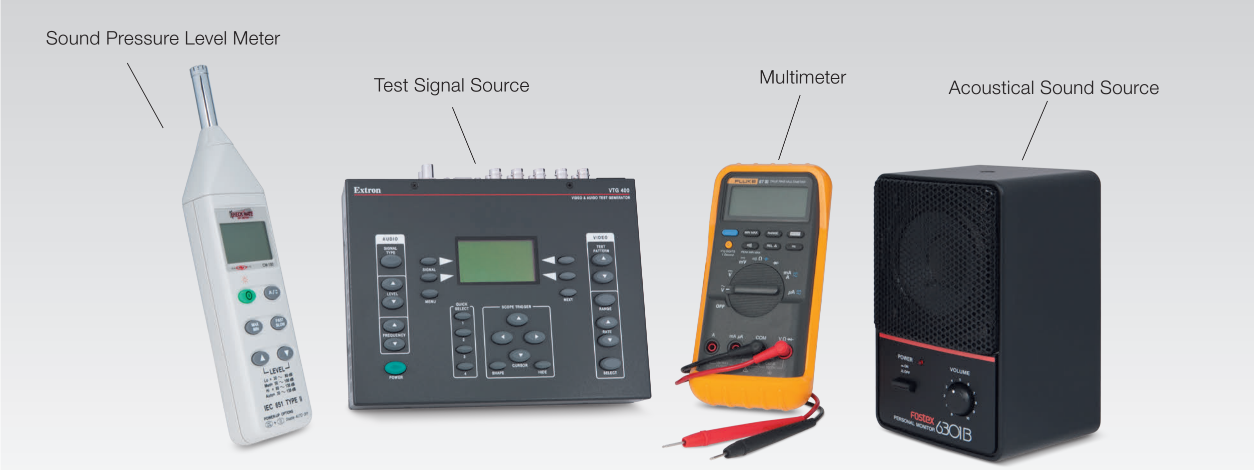
Discipline means being able to insist to everyone that you need to complete your setup, because the system isn't ready yet. It also means that the first things they are going to hear are test tones and pink noise, not the customer's favorite music selections or an awesome movie sound clip. This actually works in your favor, since there is no better way to clear a room than to play pink noise for 45 minutes straight.

Priority number two is creativity. Site conditions and project schedules may limit how much you can get done at any given point in time. Sometimes you need to be creative to get results, given the practical limitations.

## What Tools Are Needed for Setting Up Gain Structure?

Equipment for testing, measuring, and evaluating audio systems can include many types of audio devices, but here we will only list what is necessary for setting up audio system gain structure. For that, an AV technician should have the following:

- **Digital voltmeter or multimeter** – A device that measures rms voltage. Ideally, it would be rated for a 20 Hz to 20 kHz frequency response.




Proper audio gain structure requires some essential tools for making sound measurements and generating audio reference signals.

## Essential Tools for Setting System Gain Structure

When setting gain levels in an audio system, it is always important to have some equipment on hand for generating reference audio test signals and measuring their levels. This will allow for precise determination of signal levels at all stages of the system, from the signal source through the DSP, to the amplifiers and loudspeakers. The test signals and measurement tools are used to properly set nominal signal levels through the system, and ensure that ample headroom is available to prevent the possibility of clipping.

### Signal Generator

Many audio test signal generators are available, including the Extron VTG 400 Series video and audio test generators, as well as the NTi Audio Minirator devices commonly used by audio professionals. Software audio test generators are also available for iOS, Android devices, and PCs. It should be noted that these consumer devices will not output at the +4 dBu professional reference level, unless they are used with a pro audio interface.

**Essential Kit** – Economical tools for an AV technician, including a general purpose SPL meter, smartphone or tablet, multimeter, and an acoustical source such as a portable powered speaker. They are ideal for quick setup of audio systems with proper gain settings.

### Digital Voltmeter or Multimeter

A multimeter, such as the classic Fluke 87, is an absolutely essential tool in setting up audio systems. It is useful in setting rms voltage levels at the output of a test signal generator or other source, as well as in checking for proper cable connections throughout the system. A multimeter can also aid in confirming that there are no shorts at the amplifier outputs connected to the loudspeakers.

**Enhanced Kit** – These are more advanced tools for an AV engineer, including a professional SPL meter, AV signal test generator, multimeter, and an acoustical sound source. These tools enable precision in setting gain levels, as well as accuracy in establishing listening levels.

### Sound Pressure Level Meter

A sound pressure level - SPL meter measures the acoustic level in the listening environment. For gain structure setup, it is used to establish a nominal volume setting for the system, as well as a sonic reference for a human voice. Many electronic supply companies offer an affordable SPL meter, and there are many apps available for mobile devices. There are also higher performance SPL meters from Brüel & Kjær and others that conform to industry standards.

**Advanced Kit** – A senior AV engineer or audio professional typically uses this type of kit for sound system commissioning, which includes gain structure setup and sound power response optimization. Included are one or more calibrated test microphones and a laptop running audio analysis software, together with a multimeter and an acoustical sound source.

### Acoustical Sound Source

This is essentially a compact loudspeaker for reproducing pink noise test signals. It is used as a model for a human talking voice, providing a consistent, steady-state volume level for setting up microphone gain levels at the inputs of a DSP. The ATI Audio NG-1 and Whirlwind Qbox are compact, battery powered loudspeakers that include internal test signal generators. Another option is a portable studio monitor, such as the Fostex 6301B, which accepts balanced audio from a test signal generator. You can also use any other small, self-powered loudspeaker. When nothing else is available, a mobile device and its own loudspeaker, such as a smartphone or tablet can be used to quickly get a system up and running.

### Loudspeaker Impedance Meter

While not necessary for setting up gain structure, this is a very important tool to confirm proper cable connections from the amplifiers to the loudspeakers. Loudspeaker impedances should always be measured prior to powering up amplifiers for the first time, to make sure that there are no dead shorts to the amplifier outputs that could result in system downtime. The TOA ZM-104A is a commonly available impedance meter, while the NTi Audio Minirator MR-PRO signal generator includes impedance metering.

- **Sound pressure level meter** – A device that measures the level of sound in a given location.
- **Signal generator** – A device that generates audio reference signals, such as an Extron VTG 400 Series test generator.
- **Acoustical sound source, such as a portable loudspeaker** – Used to approximate the human voice for configuring microphones.

It should be noted that smartphones, tablets, and laptops can be used with specific apps to generate test signals, measure sound pressure levels, and function as a loudspeaker. The use of these devices should be considered when in a situation without proper test gear. However, while such devices are handy, they are not a true substitution for purpose-made, properly calibrated audio test equipment. The sidebar on the next page provides more detailed information on the equipment essential to properly setting up gain structure.

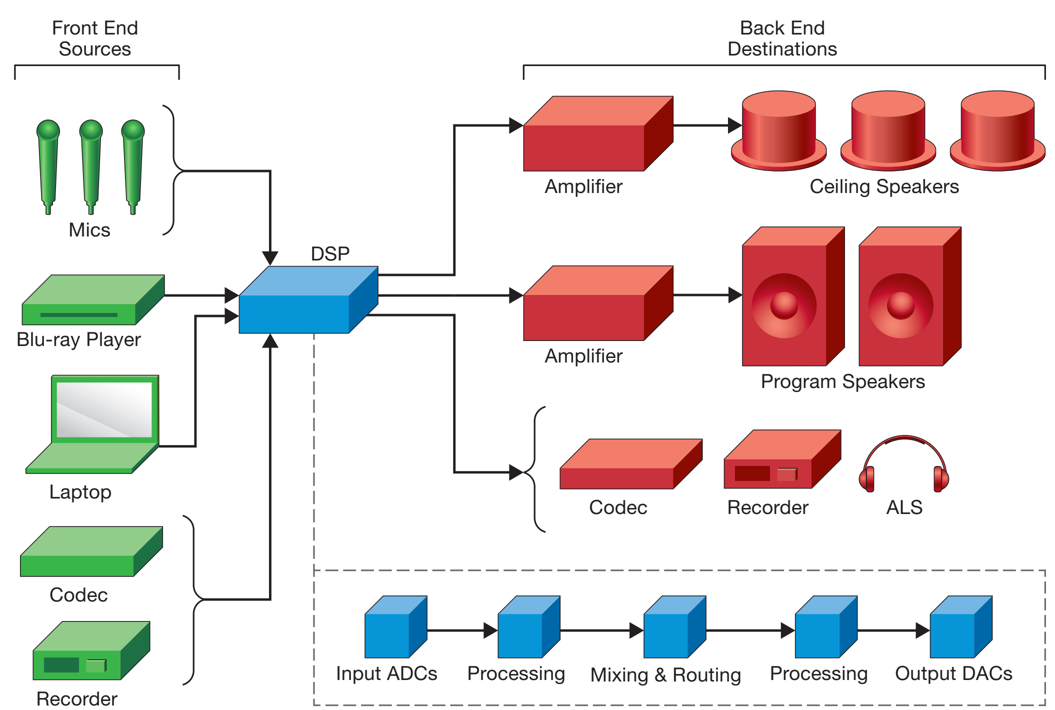
## Setting Gain Structure for Pro AV Systems

It may seem complex and overly time-consuming to properly set up an audio system. However, it really takes just four essential steps to successfully set up gain structure. We've developed a straightforward, step-by-step procedure that encompasses them and can be applied to any sound system installation in the field.

By following this procedure step-by-step, AV technicians can get an audio system up and running with very good performance. Audio professionals will find this information somewhat more simplified than the procedures they use for large-scale sound reinforcement systems. The guidelines presented here are intended to address the **practical needs and challenges** specific to pro AV applications.

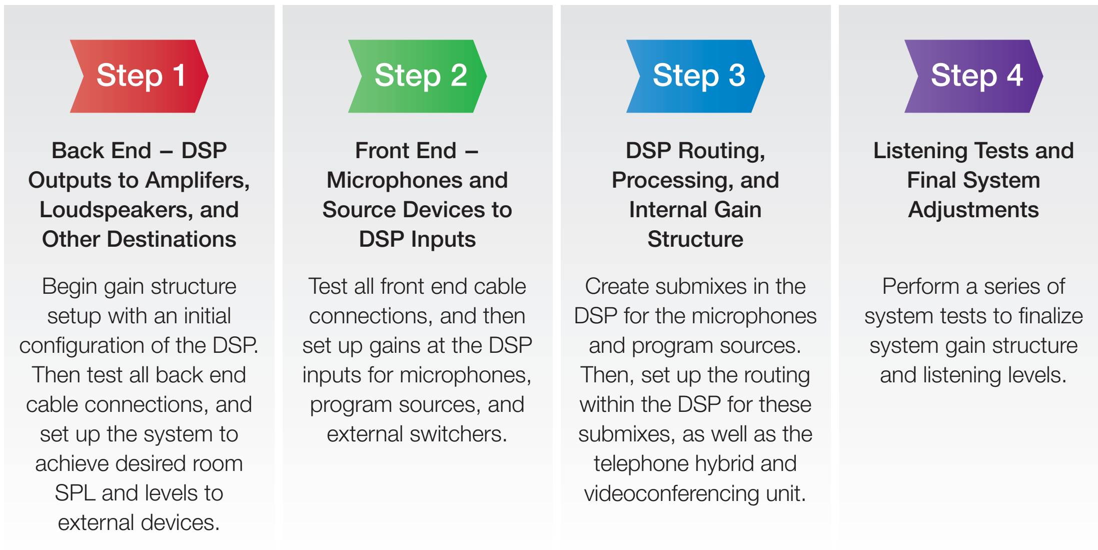
Before we proceed, let's take a moment to clarify a typical sound system in a pro AV application. Today's systems are typically based on an audio digital signal processor (DSP) that functions as the central audio device for signal routing and processing. See Figure 3. The system includes source devices such as microphones, media players, and laptop or desktop PCs, which are collectively referred to as the "front end" of the system. There are also destination devices such

Figure 3




A typical sound system in a pro AV application, including the front end – or sources, a DSP as the central audio device for signal routing and processing, and the back end – or destinations.

Figure 4




The four essential steps to setting up an audio system for proper gain structure.

as amplifiers, ceiling speakers, program speakers, and assistive listening devices (ALS), which together are known as the "back end" of the system. Other devices, including videoconferencing codecs and recorders, can exist as both source and destination. Similar principles can be applied to systems with both analog and digital-based audio signal processing and distribution.

## The Four Essential Steps

In working with pro AV and pro audio systems, usually it is best to envision the audio path from start to finish, or to begin with the sources, then the DSP, and finally the amplifiers and loudspeakers. While this sequence provides a logical signal flow, in reality it is often not practical to perform gain structure adjustments. Conflicts on the job site can delay installation of furniture, onto which microphones may be mounted and in which program sources may be installed. Gain structure setup can be expedited, however, by setting up the DSP and amplifiers already in the racks or equipment closet, while system installation is still in progress.

Therefore, the four essential steps in establishing gain structure for pro AV systems may at first seem out of order – see Figure 4. There is, however, a rationale to this approach that speeds up the process and increases the probability of a properly set up audio system.

It should be noted that before performing gain structure setup, the audio DSP program should already be written or configured, and all control system hooks defined. In complex systems, this is actually a necessity. However, as AV field technicians will attest, this is not always the case. Following the four steps will allow the field technician to develop the DSP configuration as a byproduct of the procedure.

> Control system hooks are designated functions of a DSP for interfacing with a control system. They can include gain controls, group masters, mute functions, and source selection.

## Gain Structure Procedure for Pro AV Systems

Step 1

**Back End — DSP Outputs to Amplifiers, Loudspeakers, and Other Destinations**

We will begin the gain structure setup with an initial configuration of the DSP. Then we will test all back end cable connections, and set up the system to achieve desired room SPL and levels to external devices.

### A. Initial DSP Setup

Ultimately, the goal in setting up gain structure for a DSP is to attain the best signal-to-noise ratio that the system will afford, with nominal source signals coming into and exiting from the DSP at suitable levels, and extreme signal levels handled without clipping. It is particularly important that the input analog-to-digital converters be capable of handling the full dynamic range of incoming signals without overloading. As an example, when properly set up, the Extron DMP 128 provides 17 dB of available headroom above +4 dBu nominal level, which should be more than plenty for the speech and music content typically encountered in pro AV applications.

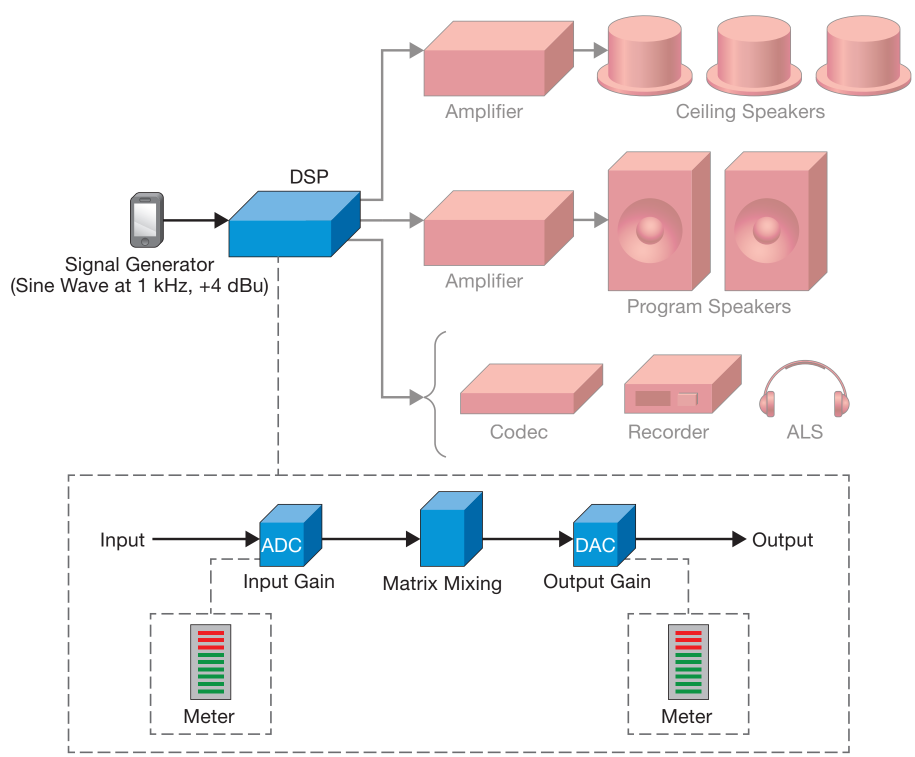
Begin the following setup after all necessary connections have been completed. This includes the DSP outputs to the amplifiers, amplifier connections to the loudspeakers, and power.

> Be sure that all amplifiers are powered off before continuing.

1. Connect your laptop to the DSP and launch the DSP application software. Load the configuration file provided to you, or begin with a new, blank configuration with the specific model of the DSP you are working with. Establish a live connection to the DSP by setting the software to the “live” or “online” mode.

   a) **If the DSP is based on a fixed processing architecture,** such as the Extron DMP 128, “push” the blank configuration to the unit, with all settings at their defaults. This includes gain settings at unity, or 0 dB, and no mixing or DSP processing activated. If a DSP configuration file has been provided for you, then open the file. Set all processing objects to bypass mode, all gain settings to unity, and mute all inputs and outputs to put the DSP into a safe status. Be sure to push the updated configuration settings to the DSP.

Figure 5




A 1 kHz, +4 dBu sine wave is used as the signal reference for confirming 0 dB or unity gain within the DSP.

| Application | Reference Signal Level | Equivalent Voltage Level |
|---|---|---|
| Consumer | -10 dBV | 316 mV rms |
| Professional | +4 dBu | 1.23 V rms |
| Broadcast | +8 dBu | 1.95 V rms |

**Table 1.** Standard audio reference levels.

b) **For a DSP based on an open architecture,** a new configuration should have all input and output levels at unity and no connections between the inputs and outputs. If you are presented with a pre-programmed DSP configuration, set gain levels throughout to unity, or 0 dB, bypass all processing objects, and mute all inputs and outputs. Push the updated configuration settings to the DSP.

2. Set up the audio signal generator for a sine wave at 1 kHz with the output level set to +4 dBu. This is the standard reference or nominal level for professional audio. If available, you can instead use the signal generator built into the DSP, and skip ahead to #4.

If the signal generator does not specify the output level in dBu, use the multimeter to measure the voltage at the output of the generator. A +4 dBu signal level is equal to 1.23 volts rms. Table 1 lists standard reference signal levels for audio.

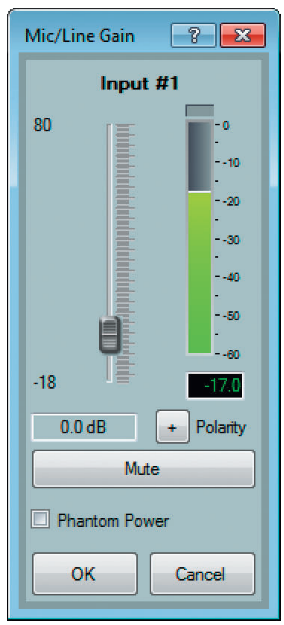



**Figure 6.** At unity gain, a +4 dBu signal reference into an Extron DMP 128 will read -17 dBFS.

3. Connect the signal generator to an input of the DSP – see Figure 5. Unmute the input. Then, through the DSP software, use the DSP input meter to measure the level of the incoming 1 kHz sine wave. With the DSP input gain at unity, the meter should be reading nominal – see Figure 6.

Some DSP models will show incoming signal levels in dBu, while Extron DSP products display them in units of dBFS (dB Full Scale). The sidebar, “Audio Signal Voltage Units” provides more information about audio voltage units including dBu and dBFS.

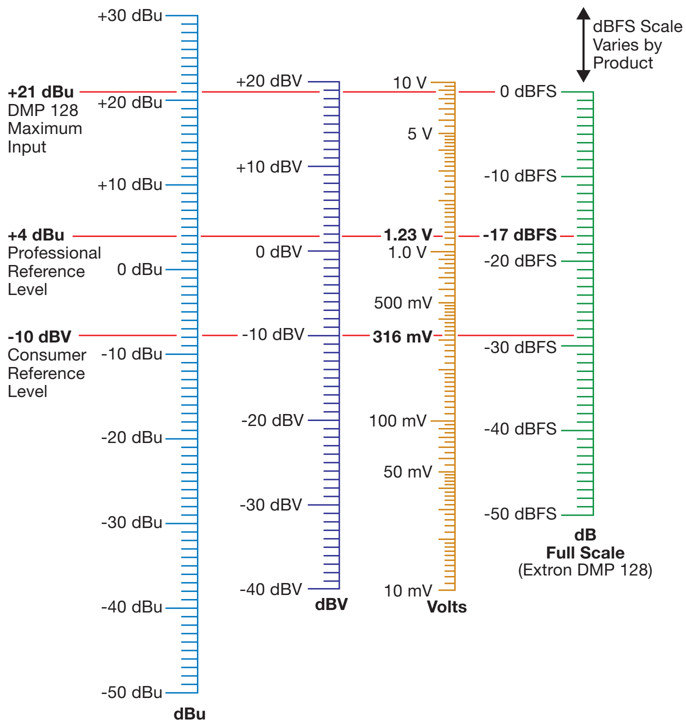
4. Route the audio signal generator through the matrix mixer to an unoccupied output of the DSP, ensuring that all routes are set for unity or 0 dB of gain. Then, unmute the output and check the meter, which should be reading the same or equivalent signal level as the generator. You can further validate proper signal level by checking for 1.23 volts rms at the unoccupied DSP output connector.

## Audio Signal Voltage Units

Measurements of audio signal levels are expressed in volts, dBu, dBV, dBFS, and a number of other units. While signal levels in decibels (dB) express the relationship between two known levels, dBu and dBV are measurement units relative to universally accepted rms voltages as zero reference points. They are relevant to audio signals in the analog domain, and are defined by the following equations:

$$dBu = 20\log\frac{V}{0.775\ volts} \qquad dBV = 20\log\frac{V}{1\ volt} \qquad dBu = dBV + 2.214$$

Expressing voltages in decibels provides a convenient means of quantifying signal gain and attenuation, signal ratios, and relating changes in signal levels to changes in listening levels. Nominal signal voltages are distinctly different between the professional (dBu) and consumer audio (dBV) worlds, with a 2.214 dB difference in the zero references for the two scales. A visual relationship between dBu, dBV, and volts can be observed in the nomograph below.




Audio professionals frequently convert between volts, dBu, and dBV. A convenient online calculator is available at www.extron.com/product/audiotools.aspx.

dBFS is a measurement unit for peak audio signals in the digital domain, with 0 dBFS being the upper threshold for digital audio signals. When connecting two audio DSP units using a digital audio pathway, 0 dBFS in one unit will equate to 0 dBFS in the other with few exceptions.

In converting from or to the analog domain, 0 dBFS represents the maximum peak signal level that can be accepted by the analog-to-digital and digital-to-analog converters. 0 dBFS is most often correlated to a device's maximum input or output level expressed in dBu. When connecting two audio DSP units using an analog audio pathway, 0 dBFS in one unit may not necessarily equate to 0 dBFS in the other. Any correlation between the devices is predicated on the maximum dBu output of the first device matching the maximum dBu input of the next device.

This is further complicated by the fact that dBFS is a peak measurement unit, while dBu is an rms measurement unit. It should be noted that there is no universal calculator for converting dBu or dBV to dBFS.

Given the difficulty of converting between these measurements, it is still useful to have a frame of reference between the analog and digital domains. In the nomograph, dBu, dBV, and voltage scales have been aligned with the dBFS scale for an Extron DMP 128, where +21 dBu is the maximum allowable input signal level at its analog-to-digital converters, and corresponds with 0 dBFS. Looking at the scales for dBFS and dBu, a +4 dBu nominal signal level would correlate with about -17 dBFS in the DMP 128. Signal meters in the DMP 128 and other Extron DSP products always provide measurements in dBFS.

For a full explanation of the dB unit in audio applications, please refer to the references listed at the end of this guide.

**B. Amplifier Cabling**

Ensure that all amplifiers are still powered off.

1. **Testing the cable connections from the DSP to the amplifiers.** Using the DSP software, create audio ties from the signal generator to the outputs connected to the amplifiers. Double-check that the audio ties you just created are all at unity or at 0 dB of gain.

   With the signal generator sending out the same 1 kHz sine wave, measure the rms voltages at the amplifier inputs to confirm proper cable connections from the DSP outputs.

   > If one signal measures about -6 dB or just 50 percent of the voltage of the others, this is likely a case where a lead has been disconnected or shorted to ground.

   Once you have confirmed cable connections into the amplifiers, turn off the signal generator.

2. **Testing for proper cable connections from the amplifiers to the loudspeakers.** This is accomplished using a loudspeaker impedance meter at the amplifier outputs. Most amplifiers have relays that disconnect the amp from the loudspeaker line when turned off. If your amplifier does not have this feature, it will be necessary to disconnect the loudspeaker circuit from the back of the amplifier in order to perform the impedance measurements.

   > Always measure the impedance of all loudspeaker circuits before turning on any amplifiers. This is very important to ensure that there are no dead shorts to the amplifier outputs. For a constant voltage system, the impedance measured should not be lower than the rated minimum impedance for the amplifier. For further information about testing loudspeaker impedances, refer to the owner's manual for your measurement device.
   >
   > If an impedance meter is not available, a DC resistance measurement can be made with a digital multimeter. While not completely accurate for determining loudspeaker impedance, a multimeter will give you a reasonable approximation, and will definitively indicate dead shorts or open circuits.

3. At this point, you have fully tested all cable connections to and from the amplifiers, ensuring that once they are turned on, there will be no damage.

**C. Amplifiers**

The next step is to establish proper gain structure at the amplifier inputs. The conventional approach is to adjust each of the amplifier input sensitivity controls, with a signal reference introduced into the system at a pre-determined level. A typical signal reference is pink noise at 10 or 15 dB below nominal. The procedure begins with all amplifier input sensitivity controls set to full attenuation. The technician then turns up an individual amplifier input control, until the desired sound pressure level is measured from the speakers. The audio is muted and the procedure repeated for each remaining amplifier input. Once setup is completed for the entire system, the amplifier input control settings are marked, often by stickers in the form of dots or arrows. This is done so that the system can be brought back to optimum operation after an amplifier replacement, or if the levels have been altered. In the days before detented gain controls, it was common practice to measure voltage gain for each amplifier channel, and include the measurements in the system verification test reports.

Technically, this procedure is the proper approach to setting amplifier input sensitivity. However, for pro AV systems that utilize audio DSP, it is more practical to fix all amplifier input gains at a predetermined level, and make individual adjustments within the audio DSP. By setting all amplifier gain controls the same, it is easy to make a quick visual inspection and detect anything out of the ordinary. With the individual output levels stored in the audio DSP, restoring the system is as easy as recalling a preset or configuration file.

A safe default setting for the amplifier input gain controls is to set them all at 12:00. While this may seem to be a random approach, 12:00 is easy to remember, easy to see in the back of the rack, and is generally 10 dB below maximum. A buffer of 10 dB helps to account for headroom, and provides a good margin of safety for amplifiers that are almost always larger than needed, and speakers that are often tapped higher than necessary.

The use of 12:00 is one of acceptable convenience. It is a case of practicality outweighing precision. Using 9:00 or 3:00 or any other setting is just as viable. What is important is to be consistent from amp to amp, and indeed, from project to project, so that anyone servicing the system can easily restore the system to nominal settings.

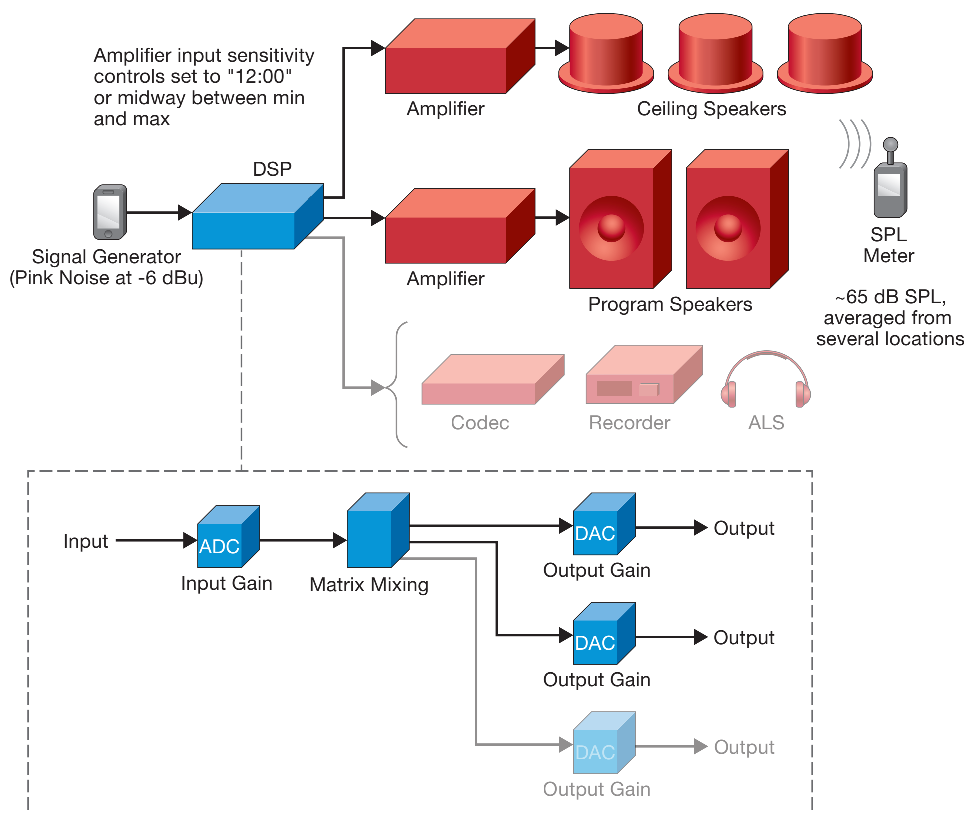
> So when is it appropriate to set the amplifier input gain above or below 12:00? When the amplifier power rating or loudspeaker tap value is larger than needed, you may find that turning the amplifier input gains lower than 12:00 keeps the noise floor down and gives you plenty of gain.
>
> When the amplifier power rating or loudspeaker tap value is smaller than needed, turning the amplifier input gains higher than 12:00 will get you sufficient gain, but at the expense of headroom.

Figure 7 illustrates the procedure for setting up the amplifiers and loudspeakers.

> If at this point you are pressed for time, and the system has only one or two loudspeaker zones, you could skip the following steps and proceed to section D.

1. Begin with the input sensitivity controls for each amplifier set at 12:00. Next, set the audio signal generator to output pink noise, and reduce the output level by 10 dB, from +4 dBu to -6 dBu, or equivalently, from 1.23 volts rms to 388 millivolts rms. For the Extron DMP 128, the meter at the input gain should hover around -27 dBFS, which is 10 dB below -17 dBFS. Attenuating the output of the generator by 10 dB will provide for a comfortable listening level once the amplifiers are turned on.

Figure 7




A pink noise test signal is used to establish a 75 dB SPL reference for the listening environment. The signal is attenuated by 10 dB from the +4 dBu pro audio reference level, so that the SPL meter reads 65 dB SPL, a more comfortable listening level for pink noise.

| Application | Reference Sound Pressure Level |
|---|---|
| Sound Reinforcement – Conference Rooms | 75 dB SPL |
| Sound Reinforcement – Auditoriums | 85 dB SPL and higher |
| Movie Sound | 85 dB SPL |
| Video Production | 78 dB SPL |
| Music Production | 78-93 dB SPL |

**Table 2.** Typical reference sound pressure levels. Some information in this table from Holman, T. (2008). “Surround Sound: Up and Running, Second Edition.” Burlington, MA: Focal Press.

If available, you can use the internal test signal generator of the DSP, instead of the external generator. Adjust the pink noise signal level within the DSP test generator module.

Reading pink noise level on a meter is notoriously difficult, due to the fact that the reading jumps around a lot. Do your best to average the result in your head and don’t try to be too precise.

2. Set up the SPL meter for A-weighting, slow response measurements. Mount the SPL meter on a tripod. If using a microphone for SPL measurements, mount it on a microphone stand. The microphone should be at ear height for the typical seated or standing listener. You can also just hold a handheld meter in position.

Refer to the SPL meter’s user manual for information on how to properly orient the SPL meter or microphone. If in doubt, point the microphone 45 degrees away from the loudspeaker.

3. Turn on all amplifiers. Select any one of the DSP outputs, pull the corresponding gain level down halfway and then unmute it. Slowly bring the gain level back up, until the pink noise seems sufficiently loud while still sounding comfortable.

At this point, the gain at the DSP output will be adjusted, so that at nominal or unity for the system, the loudspeakers are playing at a desired reference SPL. Reference SPLs for various audio and AV applications are listed in Table 2. For our purposes here, we will use a reference of 75 dB SPL, a level applicable to most pro AV sound system applications.

4. Moving the SPL meter or microphone around, measure the SPL at several locations within the loudspeaker zone, and determine the average.

5. Adjust the DSP output gain and repeat Step 4 until you measure an average SPL of about 65 dB. Recall from Figure 7 that the 65 dB SPL is equal to the 75 dB SPL reference, minus a 10 dB offset. Once you have confirmed the 65 dB average SPL, mute the DSP output.

   Note: Recall from Figure 7 that we set the signal generator to 10 dB below nominal. Our measurement would therefore be 10 dB less than the desired reference level. A 75 dB SPL reference, minus the 10 dB offset yields a 65 dB SPL target.

6. Repeat Steps 3 through 5 for the remaining outputs that feed the amplifiers.

### D. Equalization

Once the reference SPL has been established for all of the loudspeaker zones, you can now begin the process of equalizing the frequency response of the loudspeaker system. There are a number of reasons to perform system equalization, such as:

- Ensuring consistent tonal quality from one speaker zone to the next.
- Eliminating potentially annoying resonances. Resonance occurs when sound at a frequency, or band of frequencies, vibrates sympathetically with a physical attribute of the loudspeakers and/or the room.
- Helping to avoid feedback.

Equalization procedures vary and are outside the scope of this document. It is a very important topic that will be addressed in a future publication.

After equalization has been completed, adjustments for EQ and any other DSP signal processing may have boosted or attenuated levels, such that the reference SPL levels set in section C above may no longer be accurate. If time permits, it is recommended to repeat Steps 4 through 6 in section C above for each of the amplifier outputs or loudspeaker zones. Doing so ensures even coverage between loudspeaker zones after equalization.

If you are using an Extron DSP and have an installation of Extron ceiling speakers, you could apply Extron Building Blocks as a time-saving alternative to performing an equalization procedure. Building Blocks are a collection of pre-designed processor settings optimized for various input and output devices, with predefined levels, filters, equalization settings, dynamics, and more. For Extron ceiling speakers, Building Blocks provide equalization curves determined from extensive sound power response measurements of an array of nine loudspeakers of a given Extron ceiling speaker model, within an acoustically controlled environment. More information about Building Blocks can be found in the Appendix.

### E. Other Destination Devices

If other destination devices happen to be in place, such as the videoconferencing unit, audio recorder, or ALS, you can set them up at this time. Begin by turning off the amplifiers. At the DSP, create audio ties from the signal generator input to the outputs connected to these devices, and then unmute the outputs. Set the signal generator to output a 1 kHz sine wave at +4 dBu.

As an example, open the software application for the videoconferencing unit, enter into “administrator” mode if necessary, and then navigate to the audio input settings page. Using the input meter of the codec, check that the audio signal level is at the manufacturer’s recommendation for “nominal.”

- **If the codec input meter indicates no audio signal presence,** check that the input settings in the codec are correct. Also confirm that the appropriate audio DSP output is unmuted. If there is still no audio detected, check the wiring.
- **If the codec input meter is hovering at or near maximum,** verify that the codec input is set for line level, and not for microphone level. Also ensure that phantom power is turned off.
- **If the codec input meter activity appears fairly normal,** make slight adjustments to the codec input level control to get to the desired level. If necessary, make adjustments at the audio DSP output as well.

For a recording device, you may have to put it into record mode to check whether the meters are showing a nominal +4 dBu level – or 0 VU. Adjust input levels on the recording device to achieve 0 VU on its meters. If no such input level controls are available, use the appropriate outputs of the audio DSP.

If using an ALS without metering, such as a magnetic induction loop, it will be necessary to monitor the level by listening through the headphones. Set the generator to output pink noise at -6 dBu. Remember that this is +4 dBu minus 10 dB, to provide a comfortable listening level.

## Step 2 Front End — Microphones and Source Devices to DSP Inputs

In this step, we will test all front end cable connections, and then set up gains at the DSP inputs for microphones, program sources, and external switchers.

### A. Microphone Cabling

1. Confirm that all amplifiers are turned off. Set the generator for a 1 kHz sine wave at -30 dBu. The -30 dBu level roughly approximates a nominal microphone signal. Unmute all microphone inputs at the DSP.

2. At a microphone location, disconnect the microphone from the cable and connect the generator.

3. Check for a -30 dBu reading at the DSP input. Then reconnect the cable to the microphone. Repeat this procedure for all other microphones and wireless microphone base stations. Once you are finished, you have now confirmed proper cabling for all microphones in the system.

### B. Initial DSP Setup

It is essential that a high-pass filter be added to microphone inputs, for attenuating low frequency rumbles, as well as plosives – consonants such as “p” that cause a popping or thumping sound when pronounced over a microphone.

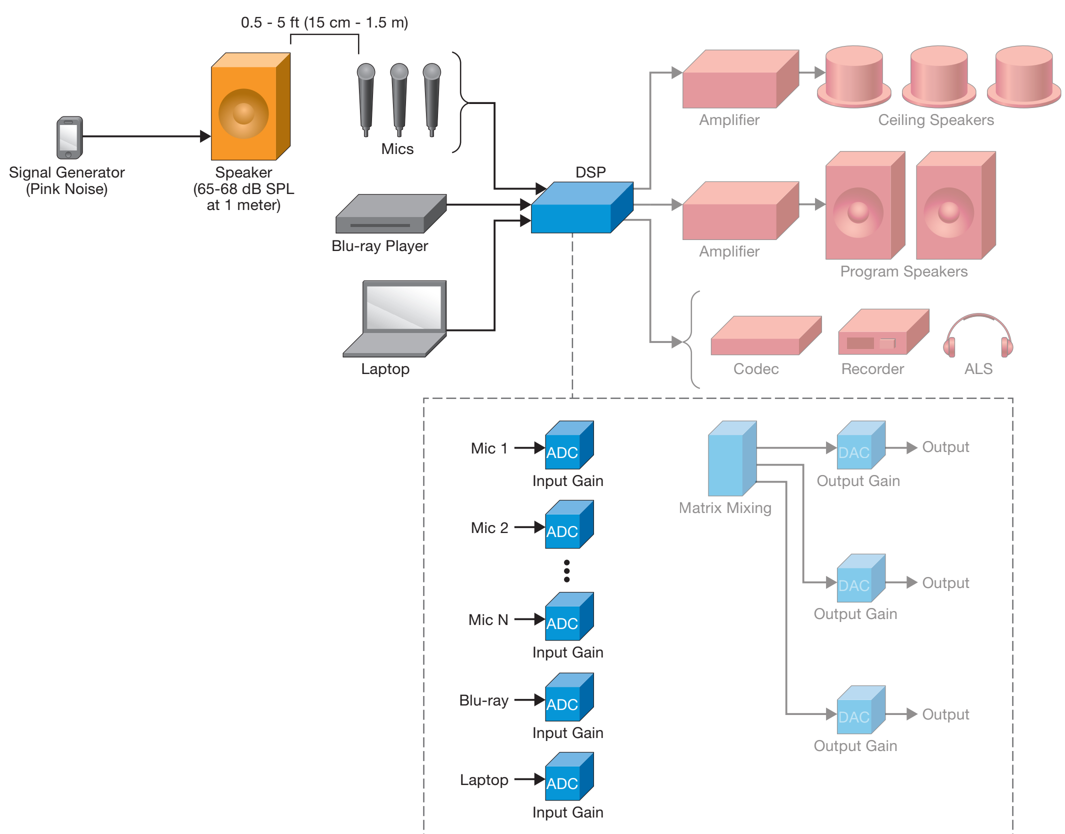
1. For each microphone input, set the high-pass filter to 100 Hz at 6 or 12 dB per octave. In rooms with significant low frequency resonances, it may be necessary to set the high-pass filtering at 150 Hz or higher.

2. If there are wireless handheld, headset, or lavaliere microphones, follow the manufacturer’s recommendations for proper setup, and then proceed with setting input gain levels in the DSP.

### C. Microphones

1. Connect the signal generator to a portable loudspeaker. Set up the generator so that it outputs pink noise from the loudspeaker at 65-68 dB SPL, when measured 1 meter away by the SPL meter – see Figure 8. This will provide a sonic reference for a raised human voice.

Figure 8




A signal generator and loudspeaker serve as a reference for a human talker, and are used to establish nominal levels for incoming mic signals to the DSP.

| Microphone Type | Distance From Talker |
|---|---|
| Gooseneck – Lectern | 1 ft (30 cm) |
| Gooseneck – Conference Table | 2 ft (60 cm) |
| Boundary – Conference Table | 2 ft (60 cm) |
| Ceiling Microphone | 5 ft (1.5 m) |
| Handheld | 0.5 ft (15 cm) |
| Wireless Lavaliere | 0.5 ft (15 cm) |
| Wireless Headset | 2 in (5 cm) |

**Table 3.** Typical talker-to-microphone distances.

Why 65-68 dB SPL? Depending upon which textbook or resource you use, standard conversational speech is defined as measuring 60-63 dB SPL at 1 meter. Similarly, a raised human voice is defined as measuring 65-68 dB SPL at 1 meter. Humans have a tendency to raise their voices when talking on the phone, as well as in meetings with a bit of heated discussion. For this reason, we have selected a raised human voice as our sonic reference.

2. Position the test loudspeaker at a microphone, using the guidelines in Table 3 for talker-to-microphone distances. If in doubt, place the loudspeaker where the talker's head would be. If the microphone is a condenser mic, be sure phantom power has been selected at the DSP input. Check the input meter and adjust the gain, so that the incoming mic signal averages around nominal, with readings between -19 dBFS to -15 dBFS for an Extron DSP, or between +2 dBu and +6 dBu if using a DSP from another manufacturer.

3. Repeat Step 2 for all microphones. Note: Steps 2 and 3 will ensure that the sensitivities of all microphones are the same with respect to a human voice.

### D. Program Sources

Setting up program sources, including Blu-ray Disc, computers, and portable media players is less straightforward than microphones. Precisely determining output levels from them can be difficult. Although the standard reference level is -10 dBV or 316 mV for consumer devices, nominal output levels often vary. The headphone outputs of laptops and PC sound cards typically deliver around 0 dBV or 1 volt nominal output. Blu-ray Disc and DVD players usually have nominal output levels of -10 dBV.

Some program sources may deliver potentially unpredictable signal levels, such as a combo VCR/DVD player, or a guest laptop connected into an open input. For these sources, consider applying automatic gain control (AGC) at the DSP input, to normalize extreme signal levels.

1. **Computers**
   When setting up a computer, open the sound card mixer and set all sources and output levels to 75%. This is not a calibrated setting, but a practical and repeatable setting that will provide an adequate signal level and a measure of safety. By limiting a user to just 25 percent additional volume on the computer, we help prevent the possibility of overloading the DSP input.

2. **Media, DVD, and Blu-ray Disc**
   There are three ways program source devices can be set up for a normalized, nominal playback level through the system:

   a) **Use calibrated, commercially available test signals recorded at the standard -20 dBFS reference level, to establish a -20 dBFS level at the DSP input.** These signals can be especially useful in setting up portable media players with variable audio outputs. In the past, test reference CDs have been available with signals recorded at -20 dBFS, but are not as easy to find today. There are also test reference DVDs and Blu-ray Discs. Internet resources are available for downloading test signals. However, beware that these signals are not necessarily calibrated to a standard reference level.

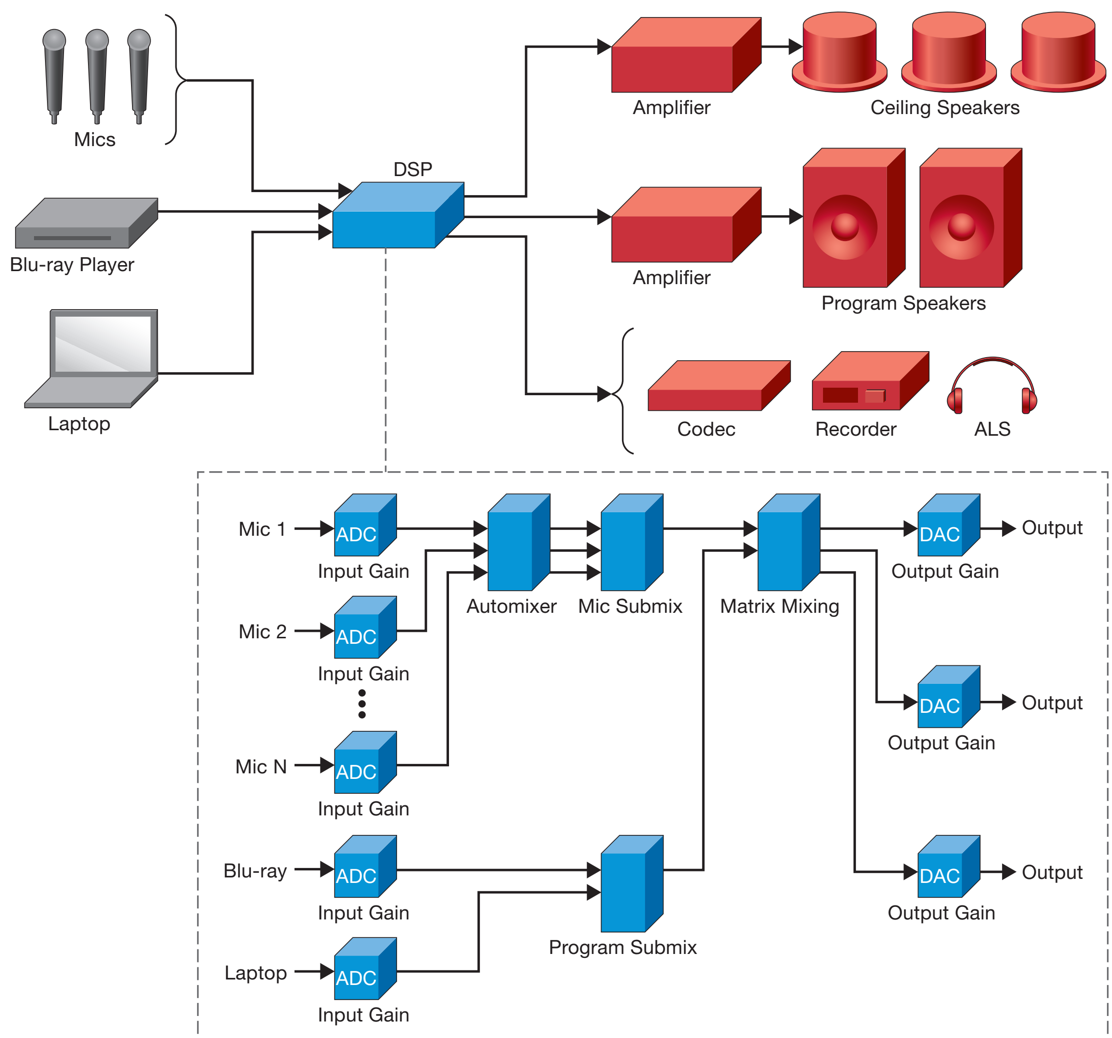
b) **Use the intended playback material from the end user.** This material can be talking or music content. It is the best for setting up system nominal level, since the content will actually be used in the application. For talking content, set the input gain so that the levels average between -20 dBFS and -15 dBFS on an Extron DSP such as the DMP 128. For music, set the gain so the meters read approximately -15 dBFS to -12 dBFS, with peaks at about -6 dBFS to ensure a safe 6 dB of headroom from the 0 dBFS clipping level.

c) **Use your own program material.** AV technicians should always have familiar audio material on hand. This should include talking content, music, and a sonic excerpt of a movie with very wide dynamic range. The movie sound clip can be useful in delivering signal level extremes to the DSP to check for clipping. If the system is intended to be used for surround sound playback, be sure that the peaks in movie content do not exceed -6 dBFS.

### E. Switchers and Matrix Switchers

In some systems, there may be video sources with analog audio, connected into a switcher or matrix switcher in front of the DSP. In these cases, set the switcher input and output gain levels to unity using the internal metering, if available, or measure the signal at the switcher output or DSP input.

Figure 9




As a starting point for establishing gain structure within the DSP, microphone and program sources should be grouped into submixes, and then routed to their respective outputs.

Set up each input source as in section D above, making sure to account for level differences between consumer and pro devices.

## Step 3 DSP Routing, Processing, and Internal Gain Structure

In this step, we will create submixes in the DSP for the microphones and program sources. Then, we will set up the routing within the DSP for these submixes, as well as the telephone hybrid and videoconferencing unit.

With nominal levels established for incoming and outgoing signals of the DSP, the next step is to bring them together into a working system by creating two submixes, one for the microphones and another for the program sources. See Figure 9. These mixes will be routed to the appropriate outputs. If present in the system, the videoconferencing far end and the telephone hybrid will also be routed to the outputs.

A comprehensive look at the “how’s” and “why’s” of configuring and programming an audio DSP is beyond the scope of this document. This is a topic that deserves in-depth discussion, and will be addressed in a future publication.

### A. Microphone Submix

1. Add automixer processing blocks to the microphone signal paths in the DSP, and configure as appropriate. If this is a system equipped for conferencing, and the DSP is equipped with acoustic echo cancellation (AEC), then add AEC processing blocks as well. As an example, a GUI screenshot of the DSP Configurator™ Software for a DMP 128 is shown in Figure 10. AEC and automixer blocks are highlighted in the areas defined by #1, and have been activated for the five microphone input paths.

> AEC is necessary for conferencing applications, and will be covered in detail in a subsequent publication.
>
> Automixing is useful in automatically managing the use of several microphones in voice reinforcement and conferencing applications. It is a topic that merits more discussion, and we will address it in a future publication.

2. At the outputs of the automixer, combine the microphone signals into a submix. For some DSP models, the submix is automatically created within the automixer. For Extron DSP products such as the DMP 128, the submix will be created by matrix mixing the signals into a virtual bus. See #2 in Figure 10.

3. Now route the submix through the matrix mixer, to the appropriate outputs of the DSP. A microphone submix typically is directed to the outputs feeding the videoconferencing codec, the telephone hybrid, and if used, the ALS and recording system. See #3 in Figure 10.

4. With the microphone submix signal path now created, you can define or incorporate a volume control point for interfacing into a control system. This volume control point will be used to provide a user control for adjusting overall microphone levels. It should be separate from the gain controls at the routing points to the outputs. Additionally, gains at the outputs should never be used for volume control, since they are already used to set amplifier input sensitivity. In our example, we will use the submix gain in Virtual Return A as the microphone volume control point, illustrated in Figure 10 as #4.

5. If local voice reinforcement is needed, the microphone submix can also be routed to the output feeding the ceiling speakers, illustrated as #5 in Figure 10. It is important to note that the ideal microphone submix level for the telephone, codec, or ALS will not be ideal for voice reinforcement. As such, the #5 matrix point will be treated differently in that it will be affected by the microphone submix volume control, but allow for final adjustment of how much voice reinforcement should be in the room. For the purpose of this article, we will call this matrix gain point the “voice lift gain control.”

> It is also viable to use a separate submix or virtual bus for the voice lift gain control. A benefit of using a separate submix is the ability to apply feedback suppression specifically to the voice lift system, without affecting the signal being sent to the far end of a conferencing session. But remember, this gain point must either exist after, or be ganged together with the microphone submix gain control, shown as #4 in Figure 10.

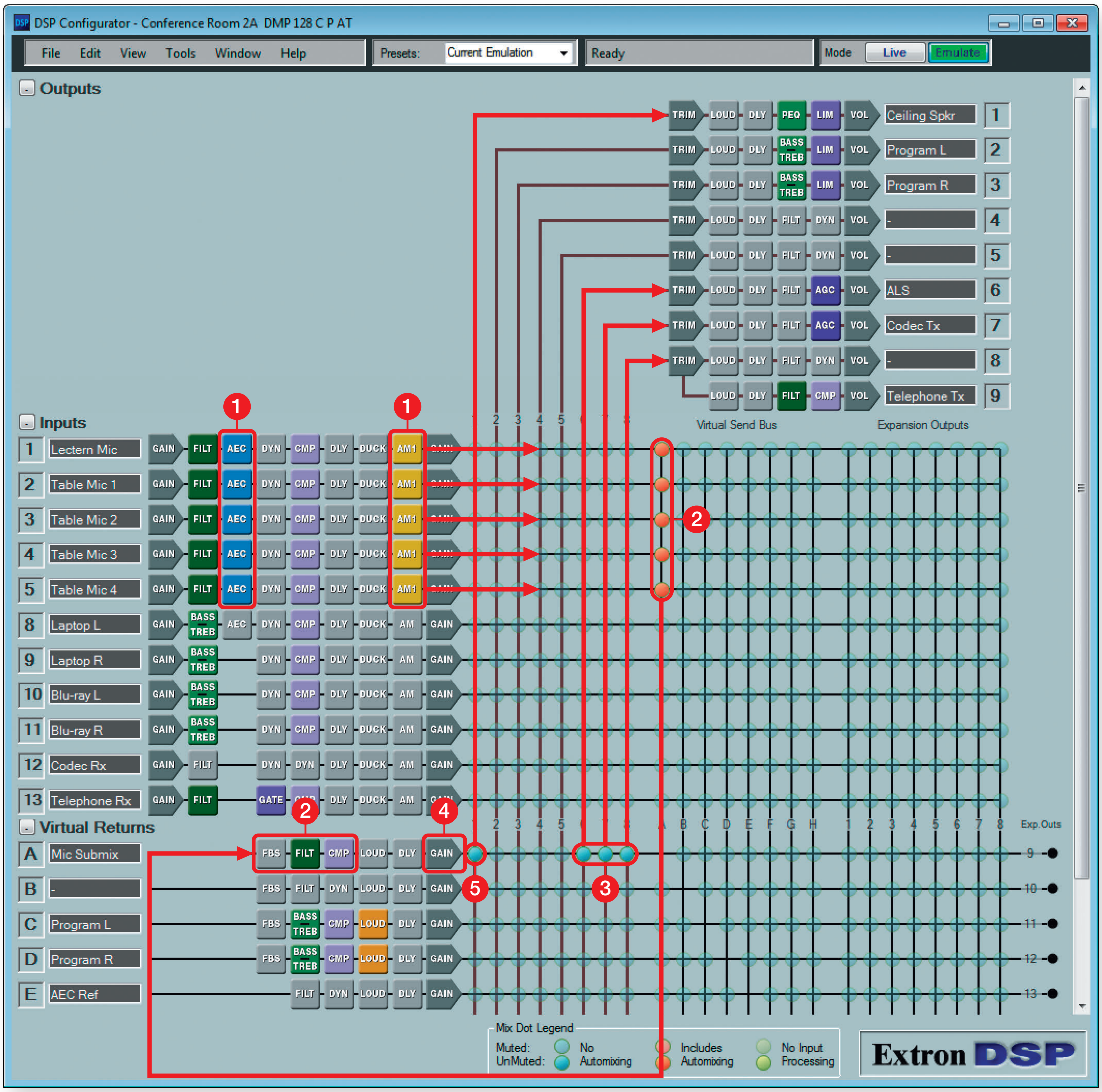



Figure 10. Mic submix and routing within the DSP Configurator Software, for an Extron DMP 128 in a conferencing application. (1) AEC and automixer processing blocks are first added to each of the five mic inputs. (2) The mic signals are routed into Virtual Return A to create a submix. (3 and 5) The submix is then routed to the outputs for the ceiling speakers, ALS, telephone hybrid, and codec. (4) A gain block is selected for a control system to provide user adjustment of microphone volume.

## B. Program Submix

1. Create a submix for the program input signal paths. For an Extron DSP such as the DMP 128, the submix will be created by matrix mixing the signals into a virtual bus. See Figure 11, #1.

2. Now route the program submix to the DSP outputs feeding the program speakers, the videoconference codec, the telephone hybrid, and if used, the ALS and recording system. This is shown as #2 in Figure 11.

3. With the program submix signal path now created, you can again define or incorporate a volume control point for interfacing into a control system. This volume control point will be used to provide a user control for adjusting the program submix level. It should be separate from the gain controls at the routing points to the outputs. As with the microphone submix, the gains at the outputs should never be used for volume control. In our example, we will use the submix gains in Virtual Returns C and D as the program volume control point, #3 in Figure 11.

## C. Telephone Hybrid and Videoconferencing Codec Inputs

1. Route the telephone hybrid to the DSP outputs feeding the ceiling speakers, the videoconference codec, and if used, the ALS and recording system – see Figure 12, #1.

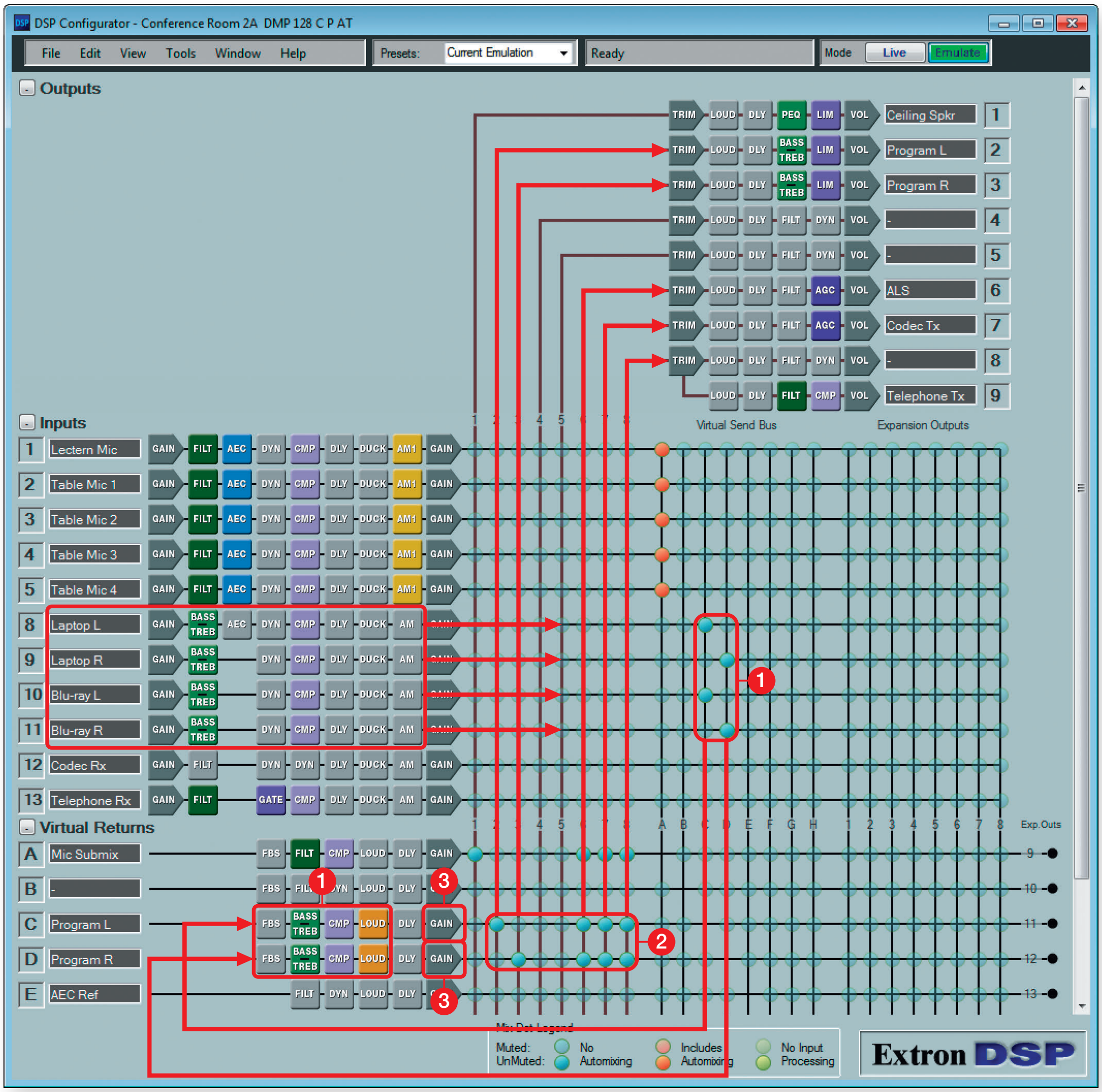


**Figure 11.** Program submix and routing. (1) The laptop and Blu-ray Disc stereo inputs are routed to Virtual Returns C and D to create a program left/right submix. (2) The submix is then routed to the outputs for the program speakers, ALS, hybrid, and codec. (3) Gain blocks are selected for a control system to provide user adjustment of program volume.

2. Route the videoconferencing codec to the DSP outputs feeding the ceiling speakers, the telephone hybrid, and if used, the ALS and recording system – see Figure 12, #2.

> ⓘ Deciding whether to put videoconferencing far end content into ceiling speakers or program speakers is rather interesting. Suffice it to say, ceiling speakers provide a more consistent level of the far end audio throughout the room, and as such, makes it easier for the AEC algorithm to do its job.

3. To establish an AEC echo reference, route the telephone hybrid, videoconference codec, and program submix to the appropriate echo reference destination. In some DSP models, this is a bus, and in other DSP products, this is a direct link to each microphone input. For our example using an Extron DMP 128, Virtual Return D will be used for the AEC echo reference – #3 in Figure 12.

> ⓘ AEC configuration will be covered in detail in a subsequent publication.

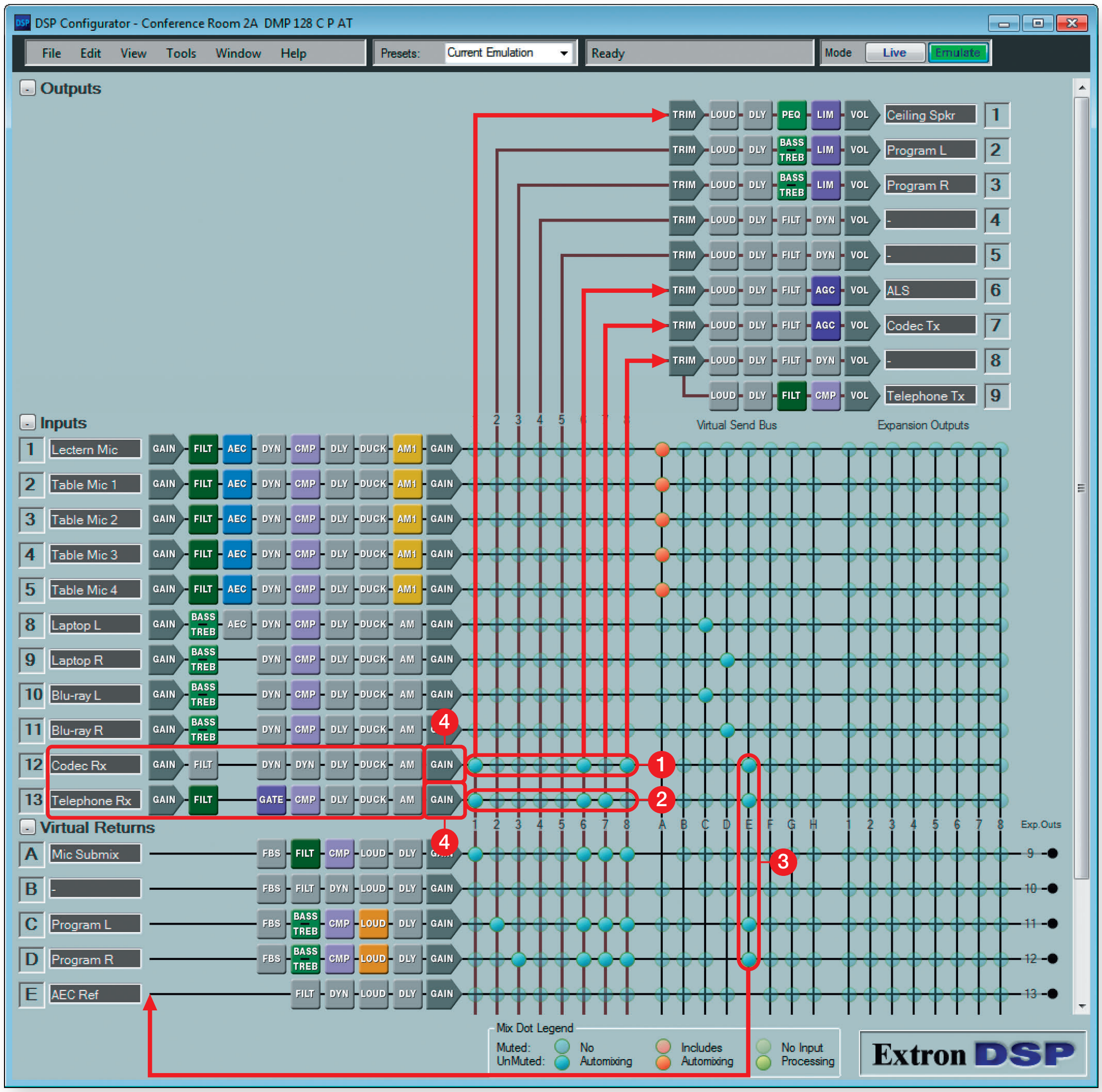



**Figure 12.** Telephone hybrid and codec routing. (1 and 2) The inputs are routed to the outputs for the ceiling speakers and ALS. The hybrid input is also routed to the codec, and likewise, the codec input is sent to the hybrid. (3) The hybrid and codec inputs are also routed with the program submix into Virtual Return E, to be used as the reference for AEC processing. (4) Gain blocks are selected for a control system to provide user adjustment of volume for the hybrid and codec far end.

4. In defining volume control points for the videoconferencing codec and the telephone hybrid, we can use either additional submixes, or we can keep things simple and use the post-processing input gain – #4 in Figure 12. These will be used for interfacing into a control system.

> ⓘ It is often the case that the volume controls within the videoconference codec will be used, due to the fact that they are user-friendly, have on-screen volume and mute indicators, and allow consistency between fully integrated and standalone videoconference systems.

Step 4

## Listening Tests and Final System Adjustments

For this final step, we will perform a series of system tests to finalize system gain structure and listening levels.

We begin by first ensuring that all microphones and program sources are feeding the outputs to the ALS, telephone hybrid, and videoconferencing codec at proper levels, before we concern ourselves with the volume levels in the room. This may seem counterintuitive, but it ensures that we have a good, solid signal level going to the outputs before we cope with the variables introduced by the loudspeaker systems, the acoustics of the room, and human perception.

In conferencing applications, it is always important to remember that there are two prime directives. One is to ensure that the audio quality of what you hear locally is at its best. The second is to ensure that communication to another site is also at its best. Both are of equal importance. When you achieve both, the meeting participants can forget that they are cities or continents apart.

**A. Microphone Inputs**

In this section, we will use the microphone inputs that we calibrated earlier, and make minor adjustments to reflect real-world conditions. It is ideal to perform this section with at least one other person serving as the talker.

If you are by yourself, you can perform these tasks by connecting the laptop running the DSP software to the display. You can then monitor signal levels on-screen as you speak into the microphones.

1. Since we do not want the voice reinforcement in the room to influence our settings just yet, we mute the voice lift gain. In the matrix mixer, turn down or mute the voice lift gain control, defined earlier in Step 3, Section A.5, and shown as #5 in Figure 10.

2. Deactivate the AEC and automixer blocks for all microphones.

3. For all table microphones, perform the following:

   a) Have someone speak at a normal talking level into each microphone and ensure that there is sufficient level at each microphone input. Observe the DSP input meter and take note of the results, but make no adjustments yet.

   b) Have someone speak loudly into each microphone and ensure that there is sufficient headroom at each microphone input. Does the microphone input clip, or does the input meter go into the red? Take note of the results, but make no adjustments yet.

   c) If any of the microphones within this group behaves differently from the rest, then troubleshoot that microphone to find out if there are wiring errors, DSP errors, or physical problems with the microphone.

   d) Once all of the microphones are behaving similarly, we can make adjustments. If the input meter levels observed from normal talking are well below +4 dBu (equivalent value of -17 dBFS in the DMP 128 – see Figure 2), then we may want to raise the input gain levels. However, if the levels observed from loud talking are approaching maximum, then we may want to lower the input gain levels. Either way, we want to adjust all microphones in this group by the same amount. Remember, we calibrated the microphones to their sensitivities in Step 2, Section C above.

4. Repeat Step 3 for all lectern microphones.

5. Repeat Step 3 for all wireless microphones.

6. Repeat Step 3 for all ceiling microphones.

Make no presumptions on how loud or soft a customer's voice might be. On one particular occasion in the author's experience with setting up audio systems, an assumption was made that a particular CEO would talk in a normal and reserved manner. It was discovered at the first event that the CEO spoke like a motivational speaker, sending his input into distortion. Needless to say, some quick adjustments had to be made.

**B. Microphone Automixing**

1. Turn on or activate the AEC and automixer for each microphone input. Open the output meter for the ALS, telephone hybrid, or codec. We will use this meter to monitor the overall voice signals being mixed. Again, AEC and automixing are beyond the scope of this document and will be covered in future publications.

2. With as many people as you can gather, have everyone position themselves at the various microphones and conduct a mock meeting. Watch the output meter to ascertain if the voice mix is adequate. Is the signal hovering around +4 dBu (-17 dBFS)? Is it too low or too high?

3. If necessary, minor gain adjustments of a couple of dB can be made at the output matrix mixing points – #3 in Figure 10. If major adjustments are needed, go back through the microphone signal path to make sure all matrix mixing points are at unity or 0 dB. If so, there may be improper settings in the automixer. Bypass or deactivate the automixer to determine whether the problem is corrected.

**C. Microphone Voice Lift Levels**

No, not yet!

**D. Program Audio**

1. At the DSP program submix for program audio – #3 in Figure 11, reduce the gain controls about halfway. At this point, the amplifiers have been powered off since Step 2, so turn them back on.

2. Play some program content that is fairly consistent in level, and bring up the program submix gain until you can hear the music a little bit.

3. Monitor the output meters to ascertain whether the program level is adequate. Is the signal hovering around +4 dBu (-17 dBFS)? Is it too low or too high?

4. Much like the microphone mix, minor adjustments of a couple dB can be made at the output matrix mixing points – #2 in Figure 11. If major adjustments are needed, go back through the program signal path to make sure all matrix mixing points are at unity or 0 dB. Also, check whether program signals are coming into the DSP inputs at a sufficient level. Make adjustments until you have a good, solid signal to the outputs.

5. Play some program content from each source, and switch between the sources to ensure that there are no major differences in level, aside from differences in content.

6. Bring up the program submix gain controls until the level sounds and feels good in the room. If desired, add some loudness processing to the program submix, to make the audio sound more full and satisfying.

**E. Conferencing Audio**

1. Turn on the telephone hybrid and ensure that you can hear the dial tone in the ceiling speakers. For the DMP 128, activate the built-in telephone hybrid within the DSP Configurator Software by loading the telephone Building Blocks, and then opening up the Phone Dialer window.

2. Using the audio tools in the videoconference codec, play whatever test tones or files are available, to ascertain that the signal level coming from the codec is adequate in the room.

3. Make a videoconference call to a far end location, and have someone there evaluate the levels of microphone and program content you are sending. Make any necessary gain adjustments at the output matrix mixing points for the microphone and program submixes – #3 in Figure 10 and #2 in Figure 11.

4. Have the far end talker speak into the system. Confirm that signals are coming into the DSP at adequate levels. Listen to the audio over the loudspeakers and adjust the gain controls at the input as needed.

5. Repeat the same process for the telephone hybrid.

**F. Microphone Voice Lift**

Now, we can bring up the voice lift gain control, or #5 in Figure 10. Why did we wait? Because using the voice lift gain control as a basis for system levels is one of the most common mistakes in setting up audio systems. It usually results in the microphone submix being too low, and the send to the videoconferencing codec turned up too high. You get the same levels in the end, but you also introduce a lot of noise.

1. In the matrix mixer, unmute and slowly turn up the voice lift gain control, not the microphone submix volume control. This is #4 in Figure 10.

2. Have someone speak into a microphone at a normal talking level. Gradually bring up the voice lift gain control until you have reached a desired listening level.

3. Listen for intelligibility. In a very live room without good acoustical treatment, reflections off of hard surfaces can compromise voice intelligibility.

> To a small degree, intelligibility problems can be dealt with using a few tricks such as:
>
> 1) Applying some tone shaping to the microphone voice lift submix, such as a boost at 2 kHz.
>
> 2) Adding an 8 kHz low-pass filter to the microphone voice lift submix.
>
> Always check signal levels after applying equalization, filtering, or other processing. The only hard and fast rule when trying to fix a room with DSP processing is: Don't expect much. The only proper way to address poor room acoustics is to incorporate acoustic treatments.

**G. Recorder and Assistive Listening System**

Check incoming signal levels for speech, program, and far end at the meters of the recorder. For the ALS, listen to the audio through the headphones. Adjust levels as necessary at the volume control for the ALS output channel.

## Final Considerations

Test the system to determine whether feedback can be introduced when adjusting the volume control points. It may be necessary to add soft limits into the DSP program to prevent a user from increasing volume to the point of feedback.

> It is recommended that a small amount of signal compression be added in the DSP, to protect against signal overload at the output digital-to-analog converters. This can be added to the inputs, the submixes, or the outputs. Remember, when it comes to compression, less is more. Set the threshold as high as possible, and the ratio as low as possible to allow system protection without affecting normal signal levels.

## Extron Building Blocks Simplify DSP Setup

For Extron DSP products, Building Blocks can simplify setup for various source devices, destinations including Extron ceiling speakers, and microphone and program submixes. Extron has created Building Blocks for many different makes, models, and types of microphones, as well as program sources including Blu-ray Disc players and PCs. Building Blocks can be used to quickly add new devices to a system, or as a starting point for further fine-tuning. More information is available in the Appendix.

## Summary

Properly setting gain levels is a very important factor in allowing audio systems to perform to their full potential. The ultimate goal is to achieve excellent sound quality that meets the end user's expectations for performance. Following a logical method for setting gain structure results in consistent performance from system to system. A system with good gain structure will often seem deceptively simple. That is its elegance.

Appendix 1:

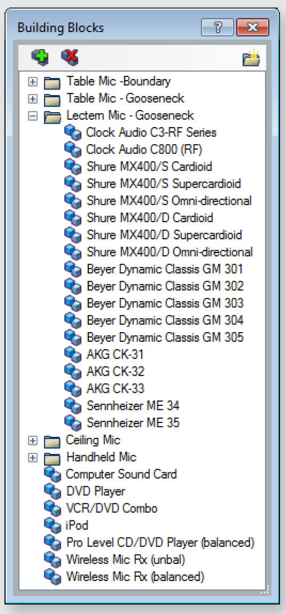
# Extron Building Blocks

Extron Building Blocks are a quick configuration tool that can significantly reduce configuration time. A Building Block is a collection of processor and gain settings for a particular portion of the signal chain. This allows configuration of an entire microphone input, line input, virtual bus return, or line output channel with just two mouse clicks.

A comprehensive set of preconfigured Building Blocks is available within the DSP Configurator Software. If you are unfamiliar with DSP processing, these Building Blocks may provide all that you need for complete configuration of your Extron DSP product.

The preconfigured Building Blocks can also be used as a starting point, providing a basic device setup while staging an installation, and a basis for fine-tuning the system during commissioning in the field. Building blocks can be modified and then saved, allowing you to create a customized set of tools unique to your needs. Alternatively, Building Blocks can be built from a blank screen or an existing configuration, allowing you total flexibility in how you create and deploy this useful and time-saving feature.

### Microphone Building Blocks




Extron Building Blocks for microphones set the gain level for a specific brand name and model, and turn on phantom power if required for that mic. A high-pass filter is inserted to compensate for common microphone problems such as low frequency noise (thumping) or "plosives" (popping). A bass and treble shelving filter is also inserted into the filter block, with gain level set to 0 dB. This allows adjustment of the microphone tone, if necessary, either via the DSP Configurator Software or by programming the gain parameters into a control system and allowing the user to make adjustments. Finally, a compressor is inserted into a dynamics processing block, applying light compression to the channel for the purpose of both normalizing diverse signal levels and protecting the system. Compressor threshold is set at -12 dBFS with a "soft knee" added and a 2:1 compression ratio. The compressor threshold is above target level so as not to affect unity gain (at the target level) through the system.

Gain levels for the microphone Building Blocks are based on several factors:

- Reference SPL level (from the talker) at 1 meter
- Distance of talker from microphone
- Microphone sensitivity (provided by the manufacturer)
- Target channel level

The table below shows different mic categories, the reference SPL level at the mic (talker level), typical talker-to-mic distance, and the target channel level. The reference SPL level accounts for a heightened level produced by a loud talker at 74 dB SPL (based on 65 to 68 dB SPL being a raised talking level at 1 meter). A loud talker level was chosen to ensure that all microphone inputs have sufficient headroom.

| Microphone Type | Reference Level | Source Distance | Target Level |
|---|---|---|---|
| Lectern – Goosenecks | 74 dB SPL | 1 foot (30 cm) | -17 dBFS (+4 dBu) |
| Table Mics – Boundary | 74 dB SPL | 2 feet (60 cm) | -17 dBFS (+4 dBu) |
| Table Mics – Goosenecks | 74 dB SPL | 2 feet (60 cm) | -17 dBFS (+4 dBu) |
| Ceiling Mics | 74 dB SPL | 5 feet (1.5 m) | -17 dBFS (+4 dBu) |
| Handheld Mics | 74 dB SPL | 0.5 feet (15 cm) | -17 dBFS (+4 dBu) |

The target channel level is +4 dBu, which allows for sufficient headroom. +4 dBu on the channel displays a meter level of -17 dBFS. Given the maximum output level of +21 dBu for the device, 0 dBFS is equal to +21 dBu. Therefore a level of -17 dBFS equals +4 dBu (+21 dBu - 17 dB = +4 dBu). When the target level is achieved at the input, maintaining unity gain through the system will produce +4 dBu at the output.

### Line Input Building Blocks

Extron Building Blocks for line level devices set the gain level for the specific device and its operating line level. Bass and treble shelving filters are inserted into the filter block, with the gain level set to 0 dB. This allows for tone adjustment, if necessary, either via the DSP Configurator Software or by programming the gain parameters into a control system and allowing the user to make adjustments. Light compression is added to the channel to normalize diverse signal levels, with threshold set at -17 dBFS and a 2:1 compression ratio. The compressor threshold is at target level so as not to greatly affect unity gain (at the target level) through the system.

The table below shows different line level device categories, and their operating line level. Gain compensation is based on the operating level and the gain required to bring the signal up to the target level. When the target level is achieved at the input, maintaining unity gain through the system will produce +4 dBu at the output. Gain compensation for line levels in dBV is calculated by first converting the source output level to dBu.

| Source Type | Operating Level | Gain Compensation | Target Level |
|---|---|---|---|
| Consumer Player | -10 dBV | +11.8 dB | -17 dBFS (+4 dBu) |
| Computer Sound Card | 0 dBV | +1.8 dB | -17 dBFS (+4 dBu) |
| Pro Level (Balanced) | +4 dBu | 0 dB | -17 dBFS (+4 dBu) |
| Broadcast Level | +8 dBu | -4 dB | -17 dBFS (+4 dBu) |

**Virtual Bus Building Blocks**

Virtual paths within the DSP are often used to create a submix, which provides a single point of control for managing a submix of microphones or program sources. Submixes can also be processed as one composite signal, as opposed to processing individual channels. The virtual bus Building Blocks provide a starting point for submix processing and control.

Virtual bus Building Blocks are provided for the following submix scenarios:

- Microphone submix
- Program channel submix

For submixing microphones, a high-pass filter is inserted, as well as bass and treble shelving filters. Light compression is added, which will help to protect both listeners and the system from many microphone channels activating at the same time.

Program channel submixes include bass and treble shelving filters, plus light compression. A loudness filter is also inserted in the path. A program channel submix allows loudness processing to be added to program material, independent of the microphones. With the two submixes separately managed, both microphones and program material can then be routed to the same output.

**Line Output Building Blocks**

The output channel Building Block inserts an equalization curve into the output signal chain of the DSP. A compressor is inserted on the output, set to a nominal level for loudspeaker protection. The output level is given a nominal setting from which you can make final adjustments to suit the room.

Line output Building Blocks are provided for program output, voice lift output, and select models of Extron ceiling speakers. The equalization curves used for the loudspeaker Building Blocks were generated by measuring the sound power response of an array of nine (9) identical loudspeakers installed in a large, acoustically dead room. For testing, 400 samples were taken throughout the sound field of the loudspeaker array and averaged together to determine the average sound power response of the nine ceiling speakers. This was repeated with the loudspeaker array at 6 feet (1.8 m), 8 feet (2.4 m), and 12 feet (3.7 m) spacing between loudspeakers. The results were merged into a single response curve for all three configurations, comprised of over 1,200 samples (total average).

*Note: An array of four (4) Extron FF 120T and FF 220T loudspeakers were used at 12 feet, 16 feet (4.9 m), and 20 feet (6.1 m) spacing, since the Flat Field Technology provides more consistent sound levels across the listening area than conventional coverage patterns.*

This total average curve was then inverted and loaded into an RTA (real-time analyzer). Calibrated pink noise was introduced into the Extron DSP, and the output of the DSP presented to the RTA. The RTA was set to A - B subtraction mode, with the total average curve subtracted from the real-time pink noise. The resulting response on the RTA was an electrical facsimile of the average acoustic response of the loudspeakers. Parametric filters were then employed on the DSP to flatten the response curve on the RTA. Those filters were then saved into the Building Block for the specific Extron ceiling speaker model.

The purpose of the loudspeaker Building Blocks is to normalize the inherently non-linear response of a ceiling speaker array, as opposed to flattening the on-axis response of a single loudspeaker. Depending on the source, additional “sweetening” of the signal may be desired. For microphones, a high-pass filter should be employed to eliminate popping and in certain cases, tone shaping of the frequency range for speech may be desired. For music sources, a bass boost is often employed to sweeten the sound at lower levels. Since the sweetening for microphones and music sources are, for all intents and purposes, somewhat opposite, adding this to the outputs would render an output ideal for one or the other, but not both. As such, Extron has included the sweetening curves into the input and virtual bus Building Blocks, allowing voice lift and music to coexist in the same loudspeaker system while maintaining attributes needed for both.

Appendix 2:

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