Your Guide to Setting System Gain

SHH: SECRETS OF THE PROS UNVEILED
Grab Your Test Meters

You are on the verge of becoming a hero to everyone on your team. This how-to guide will walk you through one of the most often overlooked - yet most important - parts of setting up audio systems.

Setting the proper gain structure for an audio system is the foundation to providing a clean, undistorted signal. Knowing how to set gain in your customer’s audio system will help it sound its best.

With the right techniques for setting gain structure and equalizing an audio system under your belt, this task will seem less daunting. And maybe a little fun too.

AV professionals who take InfoComm International® classes consistently say this information is their favorite takeaway. It has even been referred to as magic. Yeah. It’s that good. So grab your tools and let’s do this.
Methods for Setting System Gain

WHERE TO SET GAIN

There are many places (or stages) within a system to make signal level adjustments. In fact, unity gain must be set in every part of the audio signal chain.

There are several locations in an audio system where gain can be adjusted.

Setting gain structure means setting these adjustments properly so that the system gives the best performance and the user does not hear hiss, noise, or distortion from the loudspeakers.

Most mixers will produce +18 to +24 dBu output level without clipping. 10-20 dB or so of headroom will allow for an emphatic talker or loud passage in program material. This equates to the mixer delivering approximately a 0 dBu output level when the input sources are delivering their intended level.

Set the gain structure of each of these devices:

1. Microphone preamplifiers
2. Audio mixer
3. Processing devices (equalizer, compressor, limiter)
4. Preamplifiers in the mixer for microphone inputs
5. Amplifier

You do not necessarily have to set the gain structure of all of these devices in this order. You can set unity gain in your mixer first, for example, and then your microphone preamplifiers. You just need to make sure that you set gain in every item.

One exception, though, is that you should set the input of the power amplifiers last.
IMPORTANCE OF SETTING SYSTEM GAIN

Setting gain provides an optimal signal-to-noise ratio for an AV system. It also helps you avoid signal distortion.

Under normal circumstances, the audio mixer’s output level should be near the “0” mark, as indicated by the mixer’s output level meter. This allows for peaks in the signal that may exceed zero to pass through undistorted. This is usually called **distortion clipping** because the waveform appears to be “clipped off” at its peaks.

**Clipping** occurs when a system or device can no longer provide an increase in output signal with a corresponding increase in input signal. The limits of voltage swing for the device have been reached. The tops and bottoms of the signal are typically “clipped off” due to a power supply limitation.

SIGNAL LEVELS

A microphone, regardless of the type, produces a signal level called **mic level**. Mic level is a very low level signal, with only a few millivolts (abbreviated as mV to express one thousandth of a volt).

Since mic level operates at a few millivolts, it is more prone to interference. The microphone preamplifier amplifies the mic level signal to line level for routing and processing.

Line level is where all signal routing and processing is performed. Line level in a professional audio system is about 1.23 volts.

Professional line level is 1.23 V (+4 dBu), while consumer line level 0.316 V (-10 dBV). Use of an RCA or phono connector is often an indicator of a consumer level signal.

Once you have routed and processed the signal, it is sent to the power amplifier for final signal amplification up to **loudspeaker level**. The loudspeaker takes that amplified electrical signal and transduces the electrical energy into acoustical energy.

<table>
<thead>
<tr>
<th>Description</th>
<th>Voltage Level</th>
</tr>
</thead>
<tbody>
<tr>
<td>Mic Level</td>
<td>0.001 - 0.003 volts (-60 to -50 dBu)</td>
</tr>
<tr>
<td>Line Level (Professional)</td>
<td>1.23 volts (+4 dBu)</td>
</tr>
<tr>
<td>Line Level (Consumer)</td>
<td>0.316 volts (-10 dBV)</td>
</tr>
<tr>
<td>Loudspeaker Level</td>
<td>4 volts or more</td>
</tr>
</tbody>
</table>

Proper setup of system gain results in:

1. Improved signal-to-noise ratio
2. Increasing signal
3. Decreasing noise
4. Helps to eliminate any noticeable hiss coming from the system
5. Avoid distortion and clipping
6. Improved dynamic range

An oscilloscope showing a normal 1 kHz sine wave (top) and the same 1 kHz sine wave in a clipped condition (bottom).
SELECTING A METHOD OF GAIN STRUCTURE

There are two methods for setting system gain: Unity and Optimized. Both methods are relatively simple and neither requires expensive equipment.

For a typical presentation/conference room/boardroom type environment, Unity Gain will provide an adequate system signal-to-noise ratio – around 60 dB using professional audio components. If the first device shows a signal level output of 1.23 V, you should be able to measure a 1.23 V signal all the way to the power amplifier inputs.

For a more critical listening environment such as a small studio, broadcast facility, performing arts center, lecture hall, etc., the System Optimization Method will provide the optimal signal-to-noise ratio allowed by the equipment in the signal path. This method takes more time and skill to complete than the unity gain method.
TEST EQUIPMENT: QUICK OVERVIEW

Whether you’re setting unity gain or using the system optimization method, you will need at least a cable tester, signal generator, and signal measurement device.

It is always good practice to test cable assemblies before installing them. It’s quick, easy, and always more efficient to test before the installation than to troubleshoot after the fact. Not only does this help you check the connections, but it also reveals problems (like shorts) that may originate in the connectors themselves.
Setting Unity Gain

“Unity” in the context of gain structure means “same thing in, same thing out.” In other words, neither gain nor attenuation is applied to the signal. The dB or voltage level at the output of the mixer will be the same level through the rest of the signal path. The goal of unity gain is to have the same signal level entering the device as leaving the device.

The unity gain approach commonly uses a 1 kHz tone as a reference, which is measured by either a signal analyzer or a multimeter.

Here are the benefits of the unity gain method:

- It provides an adequate electronic signal-to-noise ratio in most basic audio systems.
- It’s an easy and fast method that only requires a signal generator and voltmeter.

And here are the drawbacks to using unity gain:

- Headroom and individual clipping levels vary from device to device, even between devices from the same manufacturer.
- A downstream device may clip before the output of the mixer clips, which could possibly harm components.
Recall that the goal for setting unity gain is to operate the mixer so that the mixer output meter indicates near “0” under normal operation.

Here is the procedure for setting gain with the unity gain method:

1. **Define “0.”** “0” may mean 0 VU at 1.23 V, which is also +4 dBu, the pro-audio line level reference. “0” could also mean 0 dBu (0.775 V), or even 0 dBV (1 V). To do this:
   - Set all mixer trims, faders, crosspoint gains, masters, etc. at their zero (unity) settings.
   - Configure a signal generator to output a 1 kHz sine wave at 0 dBu (0.775 V). Apply this signal to a line level input of the mixer and observe the mixer’s output meter.
   - Adjust the trim on the channel receiving the signal from the generator until the mixer’s output meter shows a “0” indication.

2. Using a signal analyzer or an RMS multimeter, measure the output signal of the mixer. Normally, you should see something near 0 dBu (0.775 V) or +4 dBu (1.23 V).

3. **Adjust the input trim** until your analyzer or multimeter indicates the closest one of these reference levels, while the mixer still indicates a “0” output level. You will use this reference signal level to set unity gain through the rest of the system.

4. **Adjust any filters and dynamic processors in the signal path so that they will not affect the reference signal.**
   - Make certain any filters, such as equalizers, are all set to zero (unity settings)
   - Set compressors and limiters to their maximum threshold settings
   - Set downward expanders to their minimum thresholds
   - During the gain setting process, you can check that the processors are set correctly by selecting and deselecting “bypass.” This confirms that the test signal is not being affected by any unintentional processing. Final settings for the dynamics processors will typically be set after the equalization process has been completed.

5. With the signal generator connected to a mixer input and the mixer output still reading “0”, connect your analyzer or multimeter to the output of the next device in the signal path. **Adjust the gain on the device** until the analyzer or multimeter shows the reference level established earlier.

Continue this practice with each device in the signal path up to the point of the power amplifier.

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**QUICK TIP: SETTING SYSTEM GAIN**

Keep these tips in mind when setting system gain:

- Finer adjustments are easiest to execute at 0 dBu, so drive the output of the mixer to 0 dBu under normal conditions.
- A system set correctly should not produce audible noise, such as hiss.
System Optimization Method

System optimization takes a different approach than the unity gain method. While the final goal isn’t changed – mixer output metering indicates around “0” under normal conditions – **optimization provides uniform headroom throughout the signal path.** It also **maximizes the electronic signal-to-noise ratio** of the audio system. This method is employed in more critical listening environments such as a small studio, broadcast facility, performing arts center, or lecture hall.

Here are the benefits for using the optimization method:

- This method provides a much more robust signal-to-noise ratio than the unity gain method. This is because the normal system’s signal level is much farther away from the device’s noise floors. The maximum signal-to-noise ratio can be achieved.
- Each line level device has the same headroom as the mixer. Specific headroom (the amount of signal allowable between 0 dBu and clipping) will vary for each device.
- In the optimization method, you find the clipping point for each device and adjust the level to just below that point. Therefore if the mixer is clipping, all the devices downstream will be clipping at the same signal level as the mixer.

Here are some drawbacks for using the optimization method:

- It requires more time and skill than the unity gain method
- Some downstream devices, such as equalizers, can be easily overloaded, so passive attenuators may be required on their inputs
- Component replacement requires system recalibration
To set gain using the system optimization method, the mixer output is driven to clipping and then set just below clipping. Once the gain has been set on the mixer, the next device is connected and its output is set the same way.

You could use an oscilloscope to look for the clipped waveform.

You could also use a real-time analyzer to observe the harmonic energy associated with a clipped fundamental frequency.

Often however, all you will need is an inexpensive piezo tweeter.
NOTES FROM THE FIELD: USING A PIEZO TWEETER

Using a piezo for the system optimization method will require using a low frequency sine wave, such as 400 Hz, as the reference frequency. While the piezo will not faithfully reproduce a 400 Hz fundamental frequency, it will easily pass the harmonic frequencies associated with clipping the 400 Hz waveform.

In other words, the piezo will be silent when the signal is a full sine wave, but it will make noise when the signal is clipped.

This is the method for setting system gain using the optimization method:

1. As with the unity method, **set all mixer trims, faders, crosspoint gains, masters, etc. at their zero (unity) settings.**

2. **Adjust any filters and dynamic processors in the signal path so that they will not affect the reference signal.**
   - Make certain any filters, such as equalizers, are all set to zero (unity settings)
   - Set compressors and limiters to their maximum threshold settings
   - Set downward expanders to their minimum thresholds
   
   During the gain setting process, you can check that the processors are set correctly by selecting and deselecting “bypass.” This confirms that the test signal is not being affected by any unintentional processing. Final settings for the dynamics processors will typically be set after the equalization process has been completed.

3. **Configure the signal generator** using a 400 Hz sine wave outputting 0 dBu.

4. **Connect the signal generator to a line level input** of the mixer.

5. **Connect the piezo to the output of the audio mixer.** Since the 400 Hz tone falls outside the passband of the piezo, it should be silent.
6. **Increase the mixer’s master output and listen.** Once you hear noise from the piezo, you have exceeded the maximum level that the mixer will output with clipping and distortion.

7. **Reduce the level until the piezo becomes quiet again.** Since the transition between an unclipped and a clipped condition is very noticeable, a clean or clipped condition is easy to hear.

8. With the mixer’s output level adjusted at just under the clipped level, **connect the piezo to the output of the next device in the signal path.**

9. **Increase that device’s gain** until a clipped condition is indicated by the piezo, and slightly reduce the level until it is quiet again.

10. **Continue this practice with each device in the signal path up to the power amplifier.**

11. Once the procedure is complete, **go back to the mixer and reduce the master output back to the unity setting.**

Since the goal is to operate the mixer near its “0” indication, the entire system has the same headroom as the mixer, and the mixer’s metering now indicates the system’s condition.

Setting Gain Structure with Audio DSPs

**Digital Signal Processors** (DSPs) can contain the functions of all the devices normally found separately within the rack of an analog system.

Examples include but are not limited to:

- Microphone preamplifiers
- Equalizers
- Compressor/Limiters
- Delays

There are two important terms when working with DSPs:

**Software** - The software provided by the DSPs manufacturer and used to configure signal levels, routing and other parameters.

**Configuration files** - These are the DSP software files that specify signal flow settings and parameters within a DSP device. Initially configured by the system designer or system programmer, these settings are fine-tuned by the installer at the jobsite for optimum performance.
Some differences between DSP and analog systems are:

1. In a DSP device, adjustments are typically made by using a computer laptop to interface with the unit. The manufacturer’s configuration software must be loaded into the computer and then the computer is connected to the DSP via an RS-232, USB, or Ethernet connection, depending on the model. The owner’s manual will indicate software requirements and connection options that are available.

2. If the unit has a front panel display, some adjustments and settings can be made through menus via the front panel interface.

3. Input signals in a DSP can be routed to either very specific outputs or multiple outputs and these routings can occur at varying signal levels. These routings and gain settings are typically made in the crosspoint matrix.

4. Gain settings follow the same pattern as an analog system - Preamplifier gain settings bring mic level up to line level and all routing and processing is performed at line level thereafter.

5. DSPs run on configuration software that is uploaded into the device. Usually, existing software configuration files are replaced (erased) when the installer uploads new configuration files. Be careful when uploading configuration files. If an existing file is found in the device, it is prudent to save the device’s existing configuration files to your computer before uploading new files.

6. After the system is tuned for optimum performance, the configuration files should be saved as well as copied over to the computer for backup and documentation purposes.

**QUICK TIP: ANALOG AND DSP VOCABULARY**

Sometimes the equivalent analog names for the preamps and channels can sometimes be different. For example, adjustments labeled “course” and “fine” can refer to the microphone preamplifier and channel fader, respectively. Like any other piece of equipment, be sure to read the manual.
Microphone Input Adjustments

Microphones are used in almost every AV system, and they are available at almost every price point, in any configuration, and for any application that you can think of. That being said, microphones will have different sensitivities. In other words, for a given input, the amount of output will vary depending on type, construction and purpose.

To make microphone adjustments, you will need:

- The microphones
- An SPL meter
- A portable noise generator

Mixer Input Adjustments

Once you’ve set system gain, it’s time to connect and adjust microphones and program sources at the mixer inputs using the preamp/trim adjustments.

There will be a wide variety of incoming signal levels into the mixer. Microphones come in at the lowest signal level, so they will require significant gain. Line level sources, however, will need little or no gain.
This is the process for adjusting microphone levels:

1. Set all channel equalizers, faders, crosspoint gains, masters, etc. to their zero (unity) settings.

2. Set each channel’s microphone preamplifier all the way down to the minimum settings. The preamplifier could be labeled micpre, trim, or gain.

3. Connect each microphone to its appropriate mic input. If the microphone connected is a condenser microphone, engage the phantom power on that channel.

4. Determine the appropriate level that each microphone is expected to receive. For normal speech applications, this is often about 65-70 dB SPL.

   Note: Singers or enthusiastic presenters will be much louder than the conversational level of 65-70 dB SPL, so compensate accordingly.

5. Position a small noise generator outputting pink noise near the microphone.

6. Using a SPL meter positioned close to the capsule of the microphone, adjust the position and/or the level of the noise generator until the SPL meter reads approximately the 65-70 dB SPL level for conversational speech.

7. With 65-70 dB SPL at the microphone, adjust the preamp for the mixer’s microphone channel until the mixer’s output metering indicates “0.”

Repeat this procedure for each microphone channel.

NOTES FROM THE FIELD: INPUT ADJUSTMENTS FOR MULTIPLE MICROPHONES

Normally, for a speech-only system or a system that utilizes an automatic microphone mixer, only one microphone is active at a time.

If you are working on a system that has multiple microphones active simultaneously, reduce the input trim for each microphone by 3 dB for each doubling of the number of simultaneously active microphones. In other words, if you expect two microphones to be active simultaneously, adjust each preamp to read -3 dB below the “0” indication on the mixer’s output. If you expect four microphones to be active simultaneously, adjust each preamp to read -6 dB below the “0” indication on the mixer’s output.
Line Input Adjustments

Besides the microphones being amplified up to line level, other sources, originating at line level, will also be found in your audio systems. For pro audio line sources, little if any, amplification will be required for routing and processing. Recall that consumer line level (0.316 V - about 12 dBu less than pro audio line level) may require some amplification.

Line levels will vary depending on the equipment. Pro-audio line level is +4 dBu, and consumer line level is -10 dBV (-7.79 dBu). While these may be the levels you would expect, the truth is that actual levels will vary depending on each particular device, as well as the type of content that is used.

Additionally, laptops will probably have a large amount of variation in their levels. Besides the variances in sound cards, users have the capability to adjust computer output levels themselves.

Here is the process for making line input adjustments:

1. **Set each channel’s preamplifier all the way down to the minimum settings.** The preamplifier could be labeled pre, trim, gain, etc.

2. If it is available individually on the channel, **disengage phantom power for any line level source.**

3. **Connect each line level device to its appropriate line input.**
   
   Often, an audio mixer will have different inputs for mic and line level. For example, the mic input may use an XLR while the line level uses a 1/4 in. input connector. Others may use the same connection type, such as a euroblock for both mic and line, but offer a software selection for mic or line. Other mixers may use a physical switch to select between mic and line. You may also see a “pad” option. This option attenuates the level going into the mixer when connecting a line level source.

4. **With the source active, adjust the preamp for the channel until the mixer’s output metering indicates “0.”**

   If possible, use the actual source and program material to be used with the system to make adjustments.

   If the actual source is unavailable, consult the manufacturer’s information to determine the expected nominal output level and use a signal generator outputting that level as a substitute.
Power Amplifier Adjustments

So far, all of the gain adjustments have occurred with the power amplifier turned off. Now it’s time to turn it on and make sure that the amplifier is sending enough power to drive the loudspeaker to the target SPL level. The power amplifier is the last component you will adjust when setting gain.

NOTES FROM THE FIELD: SYSTEM POWER SEQUENCE

Note that it is possible to damage equipment, particularly loudspeakers, by not powering the system components on or off in the correct order.

To avoid equipment damage, the following sequence should always be used:

• Begin powering items on along the signal path. You will typically start at the mixer, and end with the power amplifiers

• When powering a system off, turn off all power amplifiers, and then turn off all other electronics, working backwards along the signal chain towards the mixer.

A common misnomer is that the adjustment on the amplifier is a gain or a wattage adjustment.
In the above photo, notice that all positions less than fully clockwise are all negative numbers. The adjustments on a power amplifier would be more correctly called input attenuators. A power amplifier properly sized and specified for the application should have no problem achieving the desired SPL level when the audio mixer output meter is near the “0” indication. In other words, when driven with the proper voltage, which is somewhere between 0.775 to 1.23 V nominal, the power amplifier will produce enough voltage to drive the loudspeakers to specified levels without distortion.

The most common issue affecting sound systems is finding the input attenuators of the power amplifier all the way up (a.k.a. “wide open”) and the output of the previous devices turned down (less than unity gain). The common result is a system that produces audible hiss when it’s turned on.

Here is the process for setting input attenuators on a power amplifier:

1. Drive the mixer with normal signal levels and output metering showing near “0.”
2. Position a SPL meter at the listener location(s).
3. With the attenuators at minimum, turn the power amplifier on and adjust the attenuators until the proper SPL level, as specified by the design, is achieved.

Boosting a signal way up at one point and turning it way down at another to compensate reduces dynamic range. It also adversely affects the signal-to-noise ratio because the signal is now closer to the noise floor.
May the Gain Force Be with You

Now you’re a master of audio in training. Get out there and put you new skills to work to lay a foundation for ah-mazing, undistorted sound.

Ready to take on equalizing the system or your next challenge? InfoComm has more tricks up our sleeve waiting for you. Master the art and science of AV at infocomm.org/education or call us to set up your personal development plan, +1.703.273.7200.