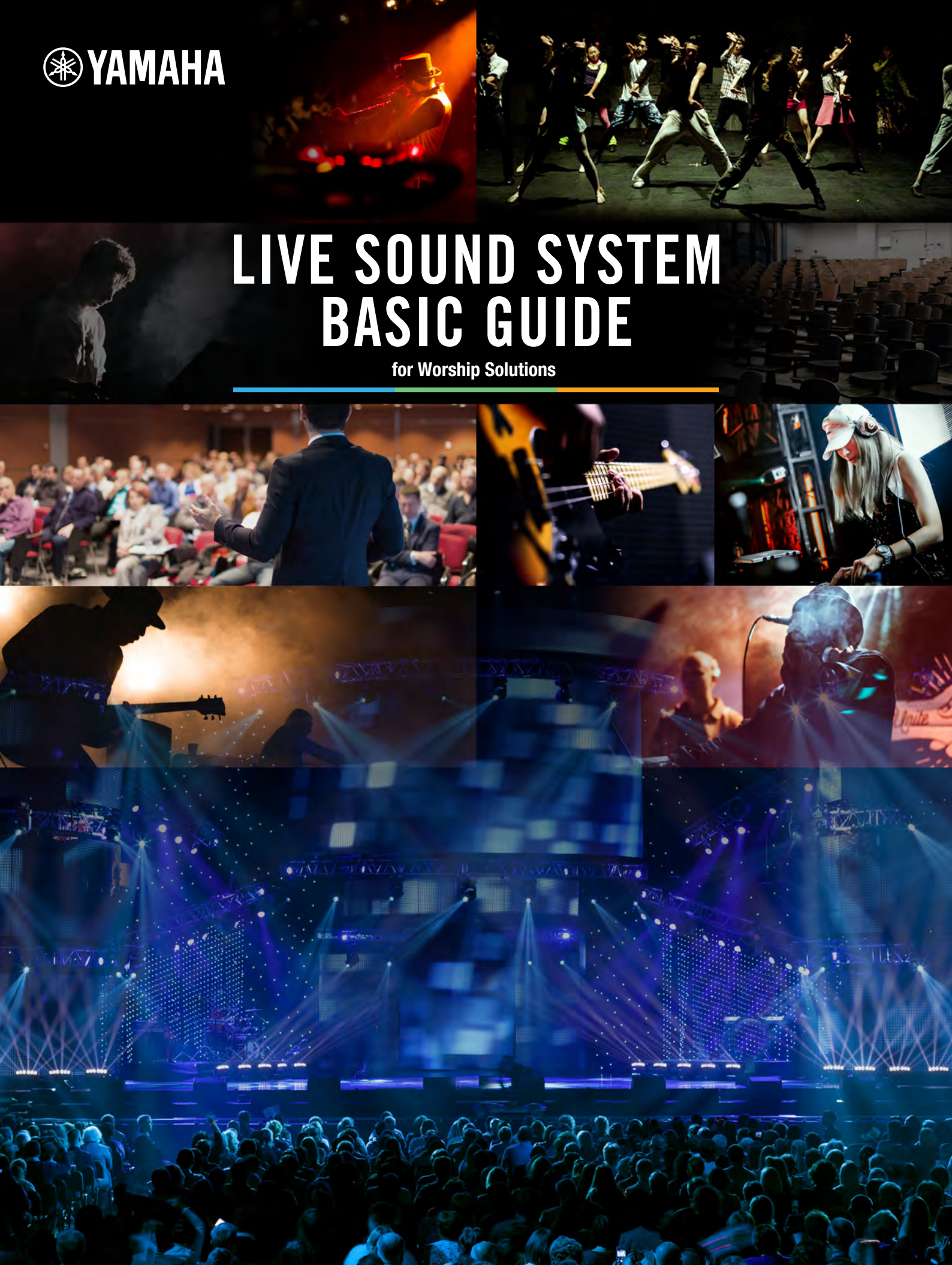




LIVE SOUND SYSTEM BASIC GUIDE

for Worship Solutions



Live Sound Basics

Setting Up

Sound for a 6-piece Band

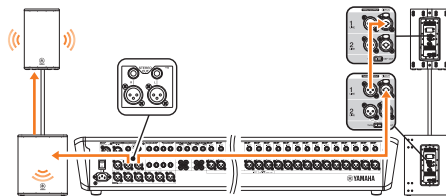
This type of system is suitable for a band consisting of drums, bass, two electric guitars, a keyboard, and vocals. We'll use an MGP32X 32-channel analog mixer, DZR12 powered speakers and DXS18XLF subwoofers for audience sound, and DXR12 powered speakers as monitors for the performers.

1. Mixer, main speaker placement, and connections

The mixer and main powered speakers are set up as follows:

- 1: Begin by positioning the mixer. The ideal location is normally in the center of the audience where the operator can hear the same sound that the audience is hearing. Place the mixer on a desk or table that is an appropriate size and height for easy operation.
- 2: Place the speakers to the left and right at the front of the stage, making sure that they do not directly face microphones (speakers that face microphones can cause acoustic feedback). Place the DXS18XLF subwoofers directly on the floor or stage, and mount the DZR12 speakers on tripod or pole stands so that the bottoms of the speakers are approximately level with audience head height.
- 3: Prepare the power for mixer and powered speakers, but don't turn the power on until all connections have been properly made.

- 4: Connect the STEREO OUTs of the mixer to the DXS18XLF INPUTs via XLR cables.
- 5: Connect the THRU/DSP outputs of the DXS18XLF subwoofers to the INPUTs of the DZR12 powered speakers via XLR cables. Set the DXS18XLF [THRU/DSP OUT] switch to [THRU].

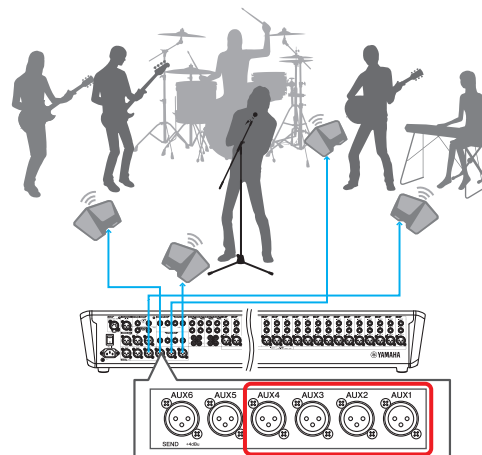


2. Performer monitor speakers

AUX send controls can be used to send different mixes to each of the monitor speakers. The AUX send monitor assignments are as follows:

- AUX 1: Vocal monitor
- AUX 2: Drum monitor
- AUX 3: Electric bass and electric guitar 1 monitor
- AUX 4: Keyboard and electric guitar 2 monitor

- 1: Place a DXR12 monitor speaker at each performer's position. Pay attention to the placement of the backing vocal microphones to prevent feedback.
- 2: Plug in the power supplies of the powered speakers used for monitoring, but don't turn the power on until all connections have been properly made.
- 3: Connect the AUX outputs from the MGP32X mixer to the monitor speaker inputs using XLR cables.



3. Microphone setup and connections

- Set up microphone stands at each vocal microphone position, and attach the microphones.
- For the electric guitars, set up the microphone stands so that the microphones point at the speaker cones in each guitar amplifier, and attach the microphones.
- For the drums, set up microphone stands for sections of the drum kit that are to be picked up (e.g. kick, snare, hi-hat, toms, overheads) and attach the microphones. Take care to avoid placing drum microphones where the drummer might accidentally hit them.
- Connect the microphones (and the D.I. Boxes if used) to the mixer inputs via XLR cables.

D.I. Box: See page 45.

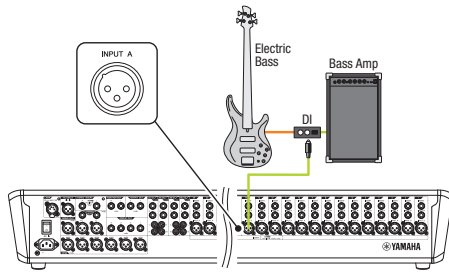
Keyboard: The keyboard can be connected directly to INPUT B on the MGP32X via phone cables if the distance is short. It may be preferable to use balanced outputs on the keyboard for longer cable runs, connecting to the mixer via TRS phone cables or XLR cables.

Note: When connecting multiple microphones and instruments, you can keep connections neat by using a snake (multicore cable) that combines multiple cables in a single jacket.

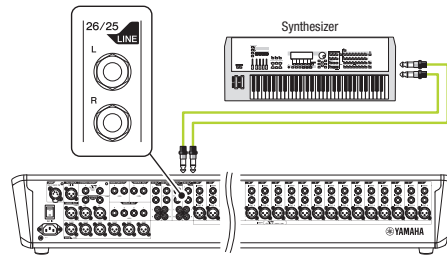


Mixer channel layout example

Channel	Input Source	
1	Kick	Dynamic microphone
2	Snare	Dynamic microphone
3	Tom 1	Dynamic microphone
4	Tom 2	Dynamic microphone
5	Floor tom	Dynamic microphone
6	High-hat	Condenser microphone
7	Drum overhead (L)	Condenser microphone
8	Drum overhead (R)	Condenser microphone
9	Electric bass	via D.I. Box
10	Electric guitar 1	Dynamic microphone
11	Electric guitar 2	Dynamic microphone
12	Keyboard (L)	via D.I. Box
13	Keyboard (R)	via D.I. Box
14	Lead Vocal	Dynamic microphone
15	Vocal (electric guitar 1)	Dynamic microphone
16	Vocal (electric guitar 2)	Dynamic microphone
17	Vocal (electric bass)	Dynamic microphone



D.I. Box: Connect the electric bass and keyboard to appropriate D.I. boxes via phone connector cables, and connect the D.I. box outputs to the mixer inputs via XLR cables. At the same time, connect the AMP or THRU connector from the D.I. Box to the bass amplifier input so that bass sound can also be heard from the bass amplifier.



Note: Keyboards, CD players, or other equipment that has stereo outputs can be connected to stereo input channels on the mixer (channels 25/26 on the MGP32X, for example), for more convenient control.

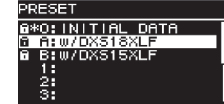
4. Power ON

- 1: Set the mixer's gain controls and faders, and all powered speaker volume controls, to minimum.
- 2: First turn the mixer power on and then the powered speakers. This power-on sequence avoids power-spike noise that can damage equipment, and possibly even your ears!

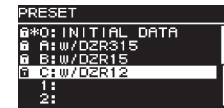
Note: Reverse the sequence when turning the power off: first turn the powered speakers off, and then the mixer.

5. Speaker presets

Recall the "w/ DXS18XLF" preset on the DZR12, and the "w/ DZR12" preset on the DXS18XLF. These are the most suitable DSP settings for this speaker system.



Recall the preset on the DZR12



Recall the preset on the DXS18XLF

6. Check sound input and output

You will need to check all channels, but here we'll provide an example for kick drum channel. The procedure for other channels is essentially the same.

Set a preliminary level for the main powered speakers. Set the DZR12 and DXS18XLF LEVEL controls to their 12 o'clock position. This is just a temporary setting that will need to be refined later.

Next, set the preliminary mixer levels.

● Turn phantom power ON for input channels connected to condenser microphones or active D.I. Boxes.

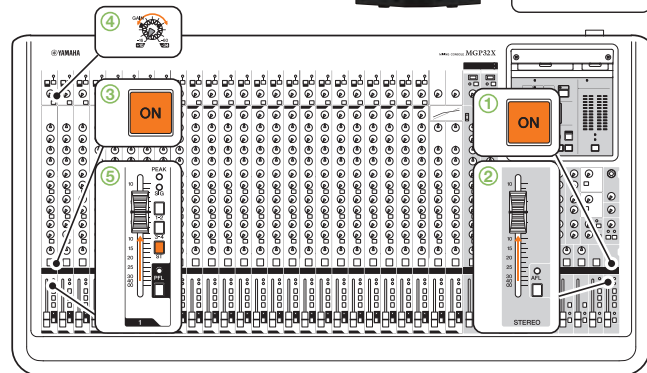
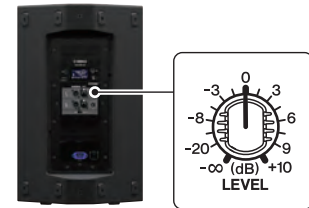
- 1: Engage the STEREO (master) channel ON switch.
- 2: Raise the STEREO (master) fader to "0".
- 3: Engage the ON switch for channel 1.
- 4: Increase the GAIN control level until the SIG (signal) lamp next to the fader lights, but the PEAK indicator does not light up even when the musicians play or the vocalist sings play at full volume.



Note: The PEAK indicator will light when the maximum input of the mixer is exceeded, possibly causing distorted sound. Set the GAIN control to a level that is just below the point where the PEAK indicator lights and distortion begins.

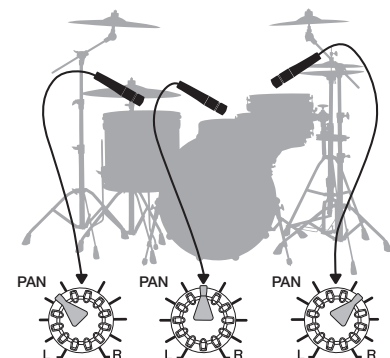
Note: Output levels from electric and electronic instruments such as keyboards can be high, and may easily exceed the mixer's maximum input level. In this case, engage the PAD switch to attenuate the input level, and then adjust the GAIN control as described above.

- 5: Engage the channel-1 ST (stereo) switch, and check for sound by gradually bringing the fader up to around the "0" mark. Repeat steps 3 through 5 for the remaining channels, adjusting the levels as necessary. The next step is to create a basic "mix" by adjusting the levels while the musicians are playing.



7. Mix levels and pan

When the input levels for each channel have been set up, have the band play and adjust the individual channel levels using the faders so that the music sounds well balanced. All parts, particularly the vocals, should be clearly audible. Pay particular attention to the balance between the vocals and louder instruments such as drums and electric guitar or bass. Use the PAN controls to adjust the stereo left-right positioning of each channel. It is common practice to pan low-frequency instruments (such as kick drums and bass guitar) and vocals (which usually play the central role in the song) to the center of the stereo sound field. In recording applications, the tom-toms are often panned from right to left, corresponding to their actual positions, so that drum fills will move dynamically in stereo. Guitars and other instruments can be panned each one to different positions for maximum clarity, matching their physical locations on the stage. However, panning instruments to one side in a live sound environment can make them less audible at audience seating on the opposite side of the venue, so many live sound engineers simply mix in mono (all channels panned to center). You will need to experiment to see what works best in each venue.



Note: The PAN control on stereo channels functions as a BAL (balance pan) control. Setting the control to its center position distributes the sound equally to the left and right. Use it to adjust the left-right balance as necessary.

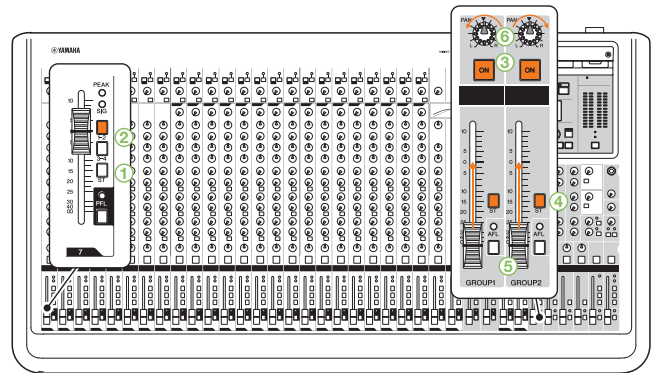
8. Grouping

The MGP32X is equipped with four GROUP buses that can be used for a variety of purposes. They are useful for grouping multiple channels prior to sending their signals to a master bus.

For example, if you have eight drum channels all eight faders must be moved simultaneously in order to raise or lower overall drum volume. If the drums are assigned to a group bus, however, this can be accomplished by using just two faders.

- 1: Disengage the drum channel ST assign switches.
- 2: Engage the drum channel GROUP 1 and 2 switches.
- 3: Engage the GROUP 1 and GROUP 2 masters.
- 4: Engage the GROUP 1 and GROUP 2 ST assign switches.
- 5: Raise the GROUP 1 and GROUP 2 faders to "0".
- 6: In order to allow stereo panning, set PAN all the way to the left for GROUP 1, and all the way to the right for GROUP 2.

Now the overall drum volume can be adjusted quickly using the GROUP 1 and GROUP 2 master faders.



9. Equalization

A general rule to keep in mind when using equalizers is that it is usually better to cut rather than boost. If boost must be applied it should be applied very gently. Consider the approximate frequency range of each individual instrument or voice, and try cutting any frequencies that fall outside their range. For example, there is very little useful low-frequency content in the sound of a hi-hat cymbal, so we may as well cut the low frequencies using the equalizer controls so that unwanted low frequencies aren't amplified by that channel. By emphasizing the characteristic registers of each instrument and attenuating unneeded content, the instruments can be more effectively isolated from one another for greater clarity and easier mixing.

This table shows examples of the approximate fundamental tonal frequency range covered by common live sound sources.

Bass Guitar	30Hz	330Hz
Kick Drum	50Hz	500Hz
Guitar	80Hz	1.3kHz
Snare Drum	100Hz	400Hz
Male Voice	100Hz	900Hz
Female Voice	250Hz	1.2kHz
Piano (88 notes)	27Hz	4.2kHz

11. Reverb

Reverb can add a sense of spatial depth and dimension to individual instruments or the overall mix. The MGP32X includes two high-performance effect processors, FX1 and FX2, providing high-quality reverb without the need for any external equipment.

In the following example we will use only FX1, but both processors can be used to create a sound rich in ambience by using different reverb programs for different channels.

- 1: Assign the HALL reverb program to FX1. The smoother PLATE program may be a better choice for vocals in some cases.

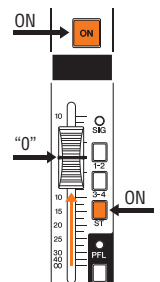


10. Compression

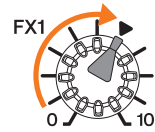
A compressor can be used to control dynamic range (the difference between the softest and loudest levels of a signal), so that the peaks and valleys are evened out. This can produce a smoother sound, make an instrument or voice more prominent in the mix, and allow the overall loudness of a mix to be increased to some extent. Compressors are often used on drums, bass, and vocals, making it easier to achieve the desired levels and "feel" within an overall mix.

As with all effects, compression should be applied carefully. Too much can make a mix sound lifeless and dull, and too much gain can increase the risk of unwanted feedback. With conventional compressors it is necessary to adjust multiple parameters (such as ratio, threshold, attack, and more) to achieve workable results. This can be difficult for inexperienced operators, and often results in a "squashed" sound with no sense of presence. The MGP32X is equipped with advanced Yamaha 1-Knob Compressors that allow anyone to achieve appropriate compression by simply turning a single knob. Turning the knob clockwise increases the compression and therefore the level of the sound, so care must be taken not to apply too much and cause acoustic feedback. Lower the channel fader a little to compensate if necessary.

- 2: Turn up the FX1 RTN control (which adjusts the reverb return level), engage the ST switch to send the signal to the stereo output, and set the fader to "0".



3: To apply reverb to a channel (the vocal channel in this example), turn the FX1 knob on the corresponding channel clockwise.



Don't overuse reverb! Too much can degrade the clarity of the sound.

After confirming that sound is output from each instrument, have the whole band play and adjust the volume of each instrument. Next, adjust the volume level of the DZR12 and the DXS18XLF to match the required sound pressure level in the venue. Take care not to make it too loud, to avoid any damage to hearing, and to avoid reaching the amplifier's limits (which may cause the sound to distort or heavily compress).

12. Monitor sends

A good monitor setup allows each performer to clearly hear his or her own sound as well as that of the other performers. Recheck the placement and connections of the monitor speakers. The speakers should not be directly aimed towards the front of a microphone, for example. Since our system uses DXR12 powered speakers for monitoring, we will switch the D-Contour switch on each speaker to "MONITOR" to optimize the frequency response for floor monitor operation.

● Monitor sends

AUX 1: vocal monitor

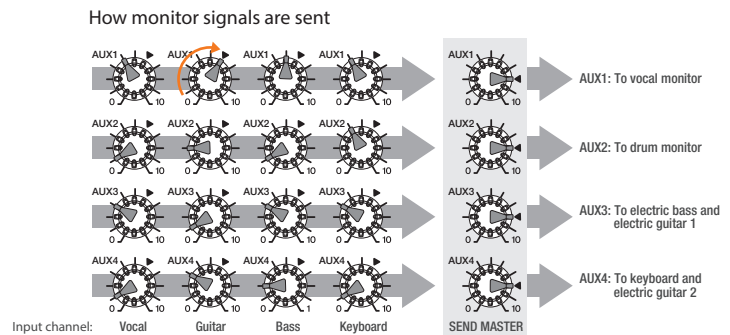
AUX 2: drum monitor

AUX 3: electric bass and electric guitar 1 monitor

AUX 4: keyboard and electric guitar 2 monitor

The relationship between each AUX send for monitors and the SEND MASTER is as shown.

The levels sent to SEND MASTER AUX 1 are adjusted by the AUX 1 knobs on each input channel.



If the vocalist using a monitor speaker connected to AUX 1 wants to hear more of the guitar sound, for example, turn the AUX 1 knob on the guitar channel clockwise to increase the guitar level in the AUX 1 monitor send. Adjust the overall level sent to the vocalist's monitor speaker by using the SEND MASTER AUX 1 knob. The monitor sends can be checked via headphones connected to the mixer by using the AFL button to the right of each SEND MASTER knob.

It is important to keep track of:

- Which channel each instrument or microphones is connected to
- Which AUX send each monitor speaker is connected to

After confirming that monitor sound is being properly delivered for each instrument, have the whole band play and adjust the levels as required. Adjust the level of each DXR12 powered speaker to achieve the required on-stage volume.

This completes basic preparations. To refine the mix, listen to how it sounds on stage and at various locations throughout the audience seating area, and make any necessary adjustments.

Related Product

Analog Mixers

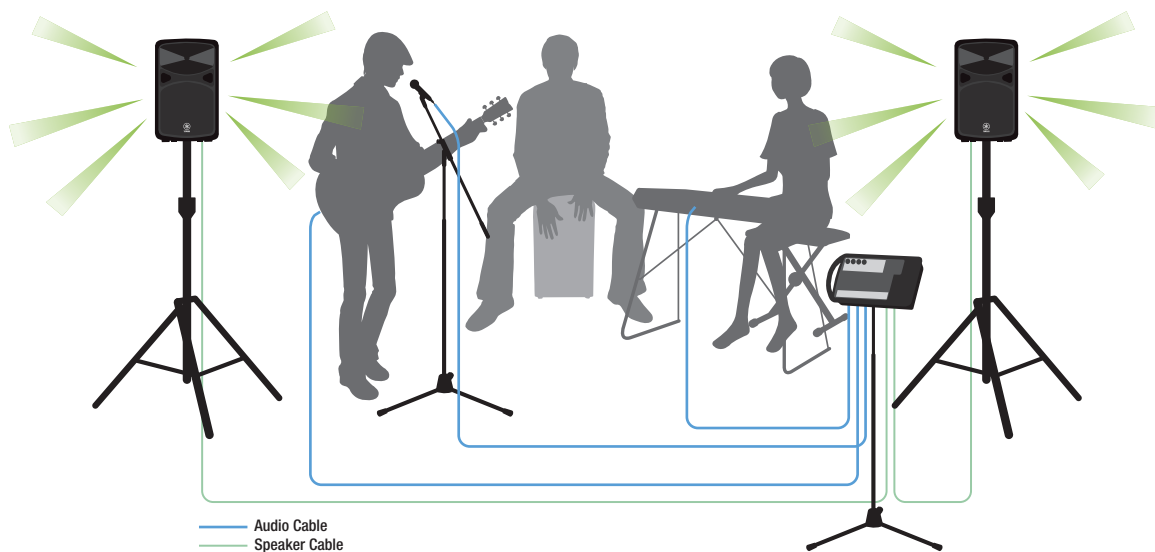


Power Speakers



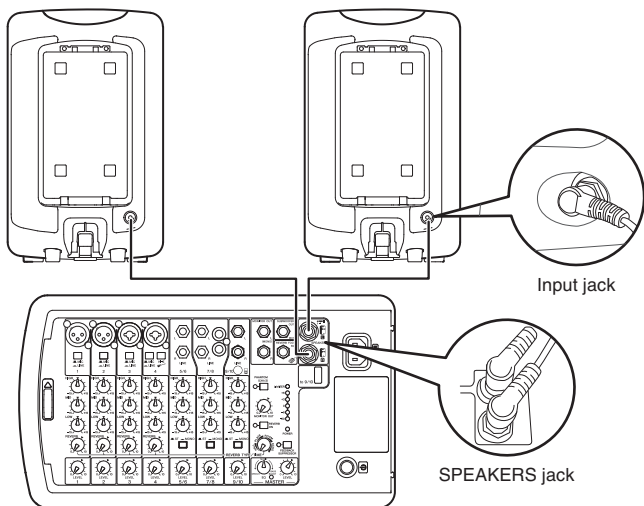
Small Ensemble

This system can provide sound for a small three-part ensemble featuring vocal, guitar and keyboard. We'll use a portable STAGEPAS 600BT sound system.



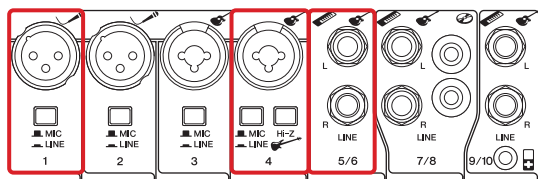
1. Prepare the power supply and connect the speakers

Don't turn power on until all connections have been made.



2. Connect the microphone and instruments to the mixer

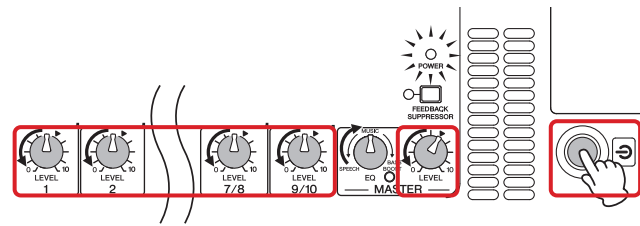
Connect the microphone to channel 1, and set the MIC/LINE switch on that channel to MIC. Connect the electric / acoustic guitar into channel 4, and set the MIC/LINE switch on that channel to LINE. Connect the keyboard to channels 5/6.



Note: The HI-Z switch on channel 4 is used when connecting a high-impedance instrument directly to the mixer without going through a DI Box. High-impedance instruments include electric guitars and basses with passive pickups. This function is provided for the phone jack input only.

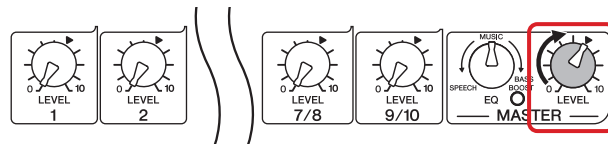
3. Turn the power on

Ensure that the MASTER LEVEL and INPUT LEVEL knobs are set to "0" (minimum), turn the power to all powered instruments on, and turn the mixer on last.

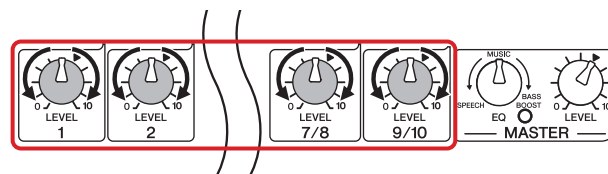


4. Adjust levels

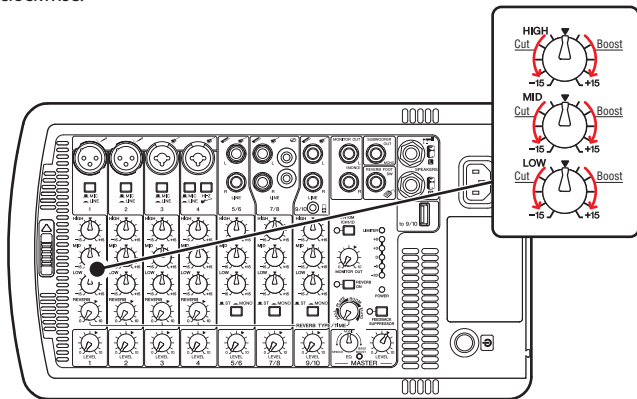
Set the MASTER LEVEL to the triangular marker position.



Adjust the volume of each channel using its respective LEVEL knob while the performers sing and play.

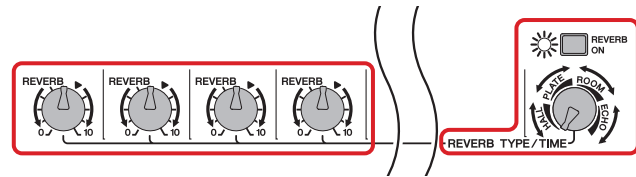


Adjust the tone of each channel using LOW, MID, and HIGH equalizer knobs. The low, mid, and high frequency bands can be boosted by turning the knobs clockwise, or attenuated by turning them anti-clockwise.



5. Add ambience with reverb

Engage the REVERB ON switch, select a reverb type, and adjust the amount of effect applied to each channel using the corresponding REVERB knobs. Reverb depth increases as the channel REVERB knobs are turned clockwise.



Types of reverberation

HALL	A reverb that simulates the reverberation of a large space such as a hall.
PLATE	A reverb that simulates the reverberation of steel plates. This gives a brighter feeling to the reverberation.
ROOM	A reverb that simulates the reverberation characteristics of a small space (room).
ECHO	An echo optimized for vocal applications.

6. Refine the mix

For this type of ensemble, the mix should normally be based on the vocals because they will probably be the most important element. Start by lowering the volume of other instruments a little, and then raise their levels as needed to support the vocals. The vocals should be clearly audible.

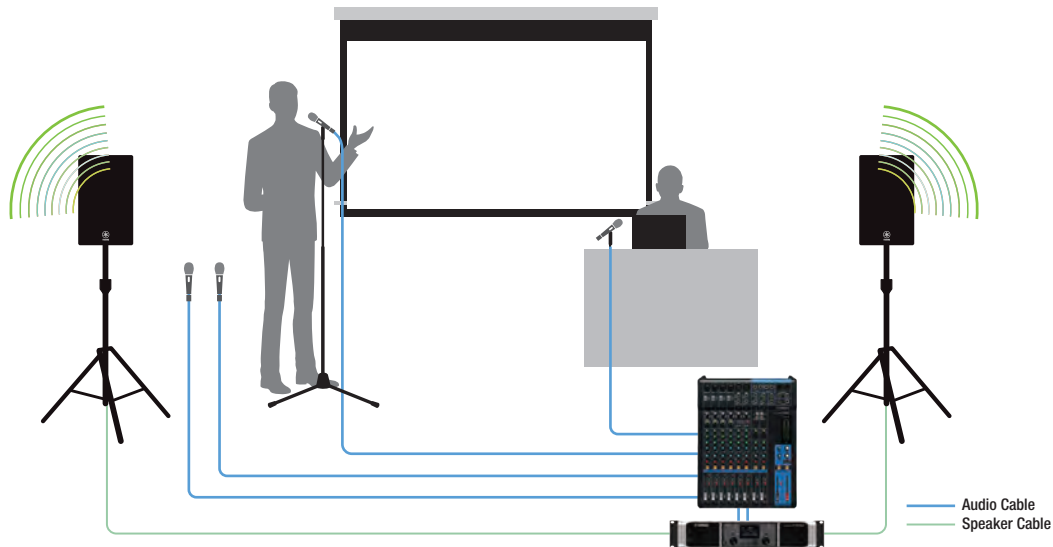
Related Product

Portable Sound Systems



Presentations

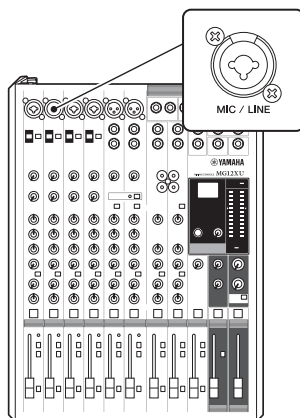
Here's a sound system that can improve the clarity and quality of communication at presentations and conferences. In this example four microphones are provided: two for speakers, and two for audience participation. A 12-channel MG12XU analog mixer, a PX3 power amplifier, and two CBR12 speakers will be used.



- 1: Begin by placing the mixer in a location where it will not interfere with the meeting. Place it on a desk or table at an appropriate height for easy access and operation.
- 2: Position the power amplifier and speakers. To prevent acoustic feedback, place the CBR12 speakers on either side of the participants, making sure that they don't point directly towards the microphones.
- 3: Prepare the AC power for the mixer and power amplifier. Do not turn the power on until all the connections have been made.
- 4: Connect the mixer and power amplifier via XLR cables. Connect the power amplifier and the speakers via speaker cables.

1. Connecting the microphones

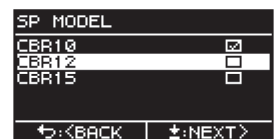
Connect the four microphones to channels 1 through 4 using XLR cables.



3. Set up the power amplifier

- 1: Select a suitable speaker configuration for the PX3. Select CBR12 in the SP MODEL page of the CONFIG WIZARD screen in the Basic mode via the front panel display.

Note: Refer the PX3 Reference Manual for details.



- 2: Start by setting the power amplifier level control to about their 12 o'clock position. This is just a temporary volume setting that will need to be readjusted later as needed.



2. Turn the power on

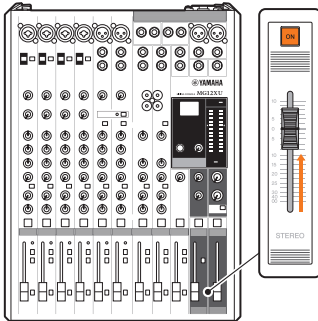
- 1: Set the mixer GAIN controls and faders, and the power amplifier level controls, to minimum.
- 2: Turn the mixer power on first, and then the amplifier power. This sequence must be followed to avoid generating startling and potentially damaging noise.

Note: Reverse the sequence when turning the power off: first turn the power amplifier off and then the mixer.

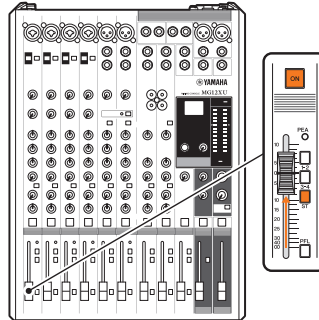
4. Adjust volume levels

The example below is for a single microphone channel.

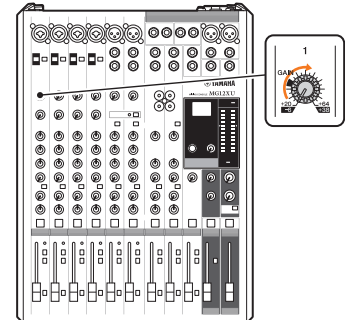
1: Engage the STEREO (master) channel switch, and raise the fader to "0".



2: Engage the channel 1 ON and ST buttons, and raise the fader to "0".



3: Adjust the GAIN to achieve the necessary volume. GAIN should be set just below the point where the fader PEAK indicator lights (the indicator should not light).



Once the basic settings for one microphone have been completed, set the other microphone channels in the same way.

5. Turn unused channels off

When one person is talking, any other active microphones will also pick up the sound. You can lower the faders on unused channels to prevent interference that can reduce clarity, but since we have already taken the time to set appropriate levels for each channel, it might be easier to use the channel ON buttons to switch unused channels off when not needed.

Related Product

Analog Mixers



Power Amplifiers

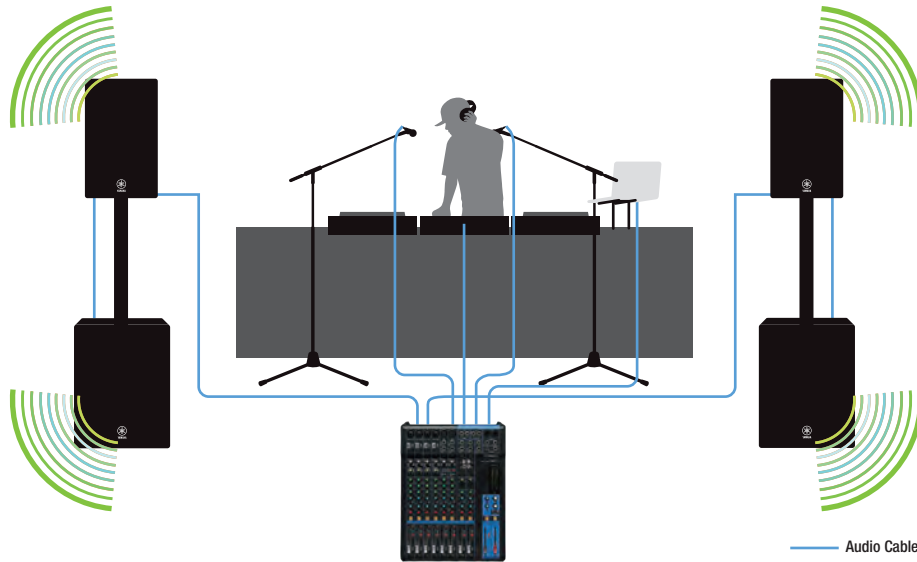


Loudspeakers



DJ Event

In this setup a DJ mixer, digital audio player (such as an iPod), and two microphones will be used for a DJ event. A 10-channel MG10XU analog mixer, DXR12 powered speakers, and DXS15mklI subwoofers will also be used.

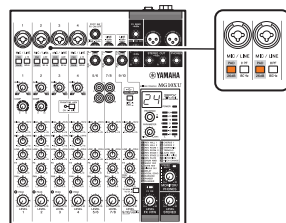


1. Equipment positioning & connections

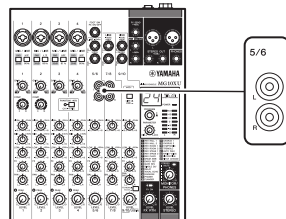
Mixer and speaker placement and connection are as follows:

- 1: Place the mixer in a secure location, ideally a spot from where the entire event space is visible.
- 2: Place the subwoofers on the floor. Mount the DXR12 speakers on tripod or pole stands so that the bottoms of the speakers are approximately at audience head height.
- 3: Prepare the AC power cords for the mixer and powered speakers. Do not turn the power on until all connections have been properly made and checked.
- 4: Connect playback devices (digital audio player, CD player, turntable, computer, etc.) to the DJ mixer.
- 5: Connect the DJ mixer outputs to the MG10XU mixer inputs. The type of connection cables required will depend on the connectors available on the DJ mixer.

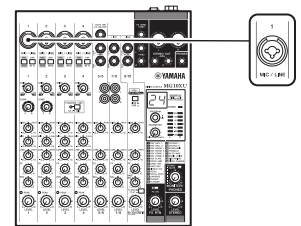
● If the DJ mixer outputs are XLR connectors: Connect XLR cables from the master outputs of the DJ mixer to MG10XU XLR inputs 3 and 4. Be sure to engage the pad switches on these inputs to avoid overload and distortion. these inputs to avoid overload or distortion.



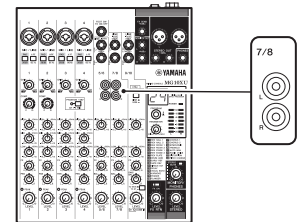
● If the DJ mixer outputs are RCA connectors: Connect RCA cables from the master outputs of the DJ mixer to MG10XU RCA inputs 5/6.



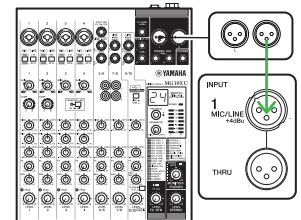
6: Connect the two microphones to MG10XU channels 1 and 2 via XLR cables.



7: Connect a playback device such as an iPod to channels 7/8 of the mixer via RCA cables.

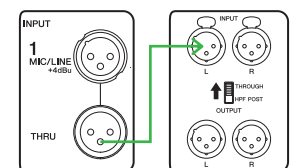


8: Connect the MG10XU STEREO OUT connectors to the DXR12 1 MIC/LINE inputs via XLR cables.



9: Connect the DXR12 THRU connectors to the DXS15mklI INPUT L connectors via XLR cables.

Note: The L and R connectors on the DXS15mklI are interchangeable. There is no difference in behavior between them.



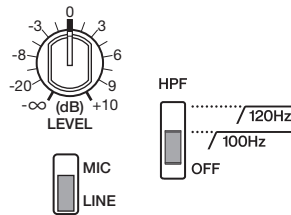
2. Turn the power on

- 1: Set the mixer GAIN controls and faders, and the power speakers/subwoofer level controls, to minimum.
- 2: Turn the mixer power on first, and then the powered speakers. This sequence must be followed to avoid generating startling and potentially damaging noise.

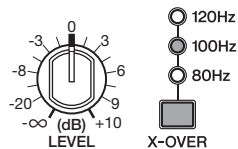
Note: Reverse the sequence when turning the power off: first turn the powered speakers off and then the mixer.

3. Sound check

1: Adjust the levels of the main DXR12 powered speakers. Set the DXR12 MIC/LINE input switches to "LINE." Set the LEVEL controls to their 12 o'clock position. Set the HPF (high-pass filter) switches to match the settings of the LPF (low-pass filter) switches on the DXS15mkII subwoofer (e.g. set both to 100Hz).

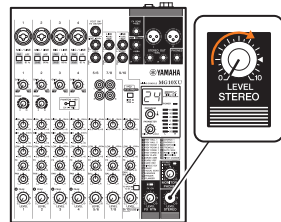


2: Adjust the level of the DXS15mkII subwoofers. Set the LEVEL controls to their 12 o'clock position. Set the DXS15mkII X-OVER switch to match the DXR12 HPF (high-pass filter) switch settings.

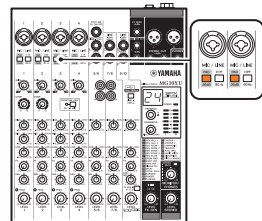


Note: These are just temporary settings for the powered speakers that may need to be readjusted later as needed.

3: Set the STEREO channel knob to the ◀ (triangular marker) position.

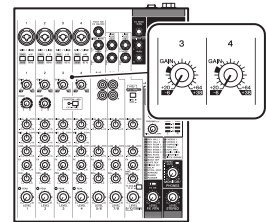


4: DJ mixers are line-level devices with high output levels that can overload the MG10XU mixer if fed to a channel that is set up for microphone input. Engage the PAD switch on channel 3/4 to lower the input level.

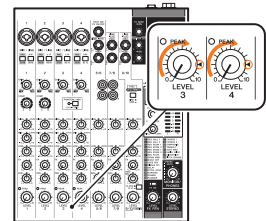


5: Adjust GAIN to a level that is just below the point at which the PEAK indicator above the LEVEL knob lights. If the PEAK indicator lights even with the PAD button depressed and GAIN set to minimum, reduce the output level of the DJ mixer.

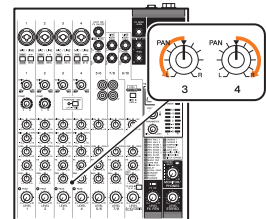
Note: Stereo channels 5/6 do not provide adjustable GAIN. If the DJ mixer is connected to those channels and its output level is too high, causing distortion, lower the output level of the DJ mixer. On the other hand, the output level of the DJ mixer can be increased if it is too low.



6: Set the channel 3 and 4 LEVEL knob to the ◀ position. (If the DJ mixer is connected to stereo channel 5/6, operate that channel.)



7: DJ mixers usually provide stereo output. Set the channel-3 PAN control to the left and the channel-4 PAN control to the right in order to produce a stereo image (the PAN settings for channels 5 / 6, if used, are preset and do not need to be adjusted). When the DJ mixer has been set up as required, set the volume levels for the microphones and iPod.



Related Product

Analog Mixers



Powered Speakers



Microphones

You have probably heard that “sound is vibrations in the air.” We are able to enjoy music because we sense vibrations in the air as sound. Microphones convert these same vibrations into electrical signals that can be amplified and processed by electronic circuitry. It is important to choose suitable microphones for your purpose.

Microphone Types

The two main types of microphones generally used for live sound applications are “dynamic” and “condenser.” Dynamic and condenser microphones capture sound in slightly different ways that can be summarized as follows:

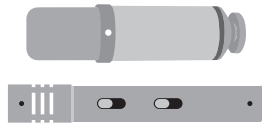
1: Dynamic microphones

Construction is simple and comparatively rugged. No power supply is required. Relatively inexpensive.



2: Condenser microphones

Good sensitivity at all frequencies. A power supply is required. Vulnerable to vibration and humidity.



Reference: Which type of microphone should I choose?

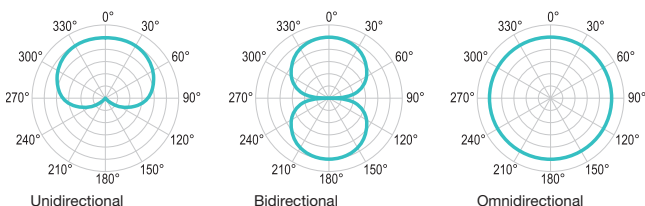
Dynamic microphones are a popular choice for general-purpose applications and can deliver excellent results. Although the “good sensitivity at all frequencies” of condenser microphones is an appealing feature, they are not always the most practical choice.

When condenser microphones are used for recording, a “pop shield” filter is required to protect against pops and noise caused by the vocalist’s breath. This is because condenser microphones are more sensitive to subtle sounds. For that reason, condenser microphones are not suited to high-volume recording. Dynamic microphones are usually preferred for situations where loud audio is to be captured. Condenser microphones are used for live sound in cases where a wide frequency range must be captured for optimum reproduction of choirs, acoustic pianos, or cymbals. They are also often used as “overhead mics” that are positioned to pick up the sound of an entire drum set.



Directivity

“Directivity” describes how well a microphone picks up sound from different directions. “Omnidirectional” microphones pick up sound equally from all directions, and will therefore pick up sound other than that from the instrument they are aimed at. In live sound situations multiple instruments will often be played on the stage at the same time, so directivity becomes important. Some microphones have switches that allow selection of different directivity patterns.



1: Unidirectional

This type of microphone, also known as “cardioid” because it usually has a heart-shaped directivity pattern, is most sensitive to sound arriving on-axis from the front. In live sound systems cardioid microphones are commonly used for vocals and instruments because they can effectively reject extraneous noise arriving from other directions. In the diagrams below, 0° corresponds the front of the microphone.

2: Bidirectional (or Figure Eight)

This type of microphone is sensitive to sound arriving from both the front and rear. Bidirectional microphones are rarely used for live sound, but they are often used for radio conversations or interviews where they can simultaneously capture the voices of two people talking face-to-face.

3: Omnidirectional

As mentioned above, this type of microphone picks up sound from all directions. Omnidirectional microphones are a good choice when it is necessary to pick up audio from a wide area. They are sometimes used for choirs, orchestras where they have to capture many instruments simultaneously, and to pick up audio from an entire stage.

Reference: What directivity should I use?

Unidirectional microphones are generally used for vocal and instrument pickup in live sound systems. Omnidirectional microphones should only be used when it is necessary to simultaneously pick up vocal or instrumental performances from multiple people, or to pick up sound from the entire stage area. Caution is required when positioning omnidirectional microphones and setting their levels, because they will pick up sound from the main and monitor speakers, potentially causing feedback.

Tip: Don’t cover the microphone grille with your hand when feedback occurs

When squealing feedback does occur, some people instinctively cover the microphone with their hand. Doing so, however, can actually make the feedback worse. Covering a unidirectional microphone makes it behave more like an omnidirectional microphone that will pick up more sound from a wider area, including any speakers in the vicinity, resulting in more feedback. Some people like to hold the microphone by its grille, but this can also increase the risk of feedback.



Other variations

Wireless microphones, lapel microphones, and headset microphones provide practical solutions to issues often encountered in live sound applications.

1: Wireless microphones

Wireless microphones can be moved around freely, without cables getting in the way. A radio frequency transmitter and receiver are used to send audio from the microphone to the sound system. Handheld wireless microphones normally contain a built-in transmitter, whereas headset and clip-on microphones often use a belt-pack type transmitter that is connected to the microphone unit via a cable. Both transmitter types are battery powered (dry-cell or rechargeable), so it is necessary to monitor battery usage to ensure that there always enough charge available. The microphone transmitter and receiver must be set to the same channel, and when multiple wireless microphones are used each one must be set to a different channel.



2: Headsets and lapel microphones

Headset microphones are, as the name implies, worn on the head, while lapel or “lavalier” microphones are clipped to the user’s lapel or other area of clothing that is not too far from the mouth. The advantages of both types are that they are relatively discreet and leave both of the user’s hands free. Headset types are often used by performers who sing and dance, while lapel microphones are used by television presenters and interviewees. Headset and lapel microphones are sometimes wired, but most commonly used wirelessly.



Note: Many headset and lapel microphones are omnidirectional, so caution is required when using them with a live sound system. Unidirectional headset microphones can reduce the risk of feedback

Cables and Connectors

Various types of cables and connectors are used to link live sound system equipment. Appropriate cables must be used to ensure proper operation and optimum performance. In some cases, the wrong cable can be dangerous, so cables must always be selected and used with care.

Connectors for Live Sound

XLR Connectors



XLR connectors are frequently used in professional applications where reliability is essential. They provide compatibility with balanced lines, described later, are rugged and resistant to deformation, and feature a locking mechanism that prevents accidental disconnection. They are used in live sound systems for microphones, power amplifiers, powered speakers, and more.

Phone connectors



There are two main types of phone connector: stereo and mono. Stereo phone connectors, also referred to as TRS (Tip-Ring-Sleeve) connectors, are used to transfer stereo signals and insert effects. TRS phone connectors can also be used for balanced connections. Mono phone connectors are used primarily to make unbalanced connections for instruments such as electric guitars and keyboards, as well as some low-power amplifiers and speakers.

Tip: Another type of connector, referred to as a “combo jack,” can be used with XLR or phone connectors. This type of connector is sometimes found on mixers or audio interfaces.



Cables for Live Sound

Audio signals can be transferred between different devices in a live sound in either unbalanced or balanced form. These are different transmission methods that require different types of cable.

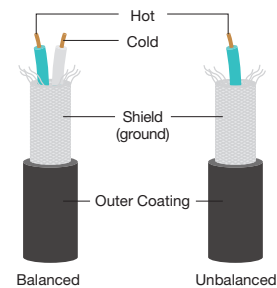
The characteristics of balanced and unbalanced signals are:

Balanced: excellent noise and interference rejection

Unbalanced: simple but prone to picking up noise and interference

Balanced lines carry a two-phase signal via cables that have three conductors: hot, cold, and ground.

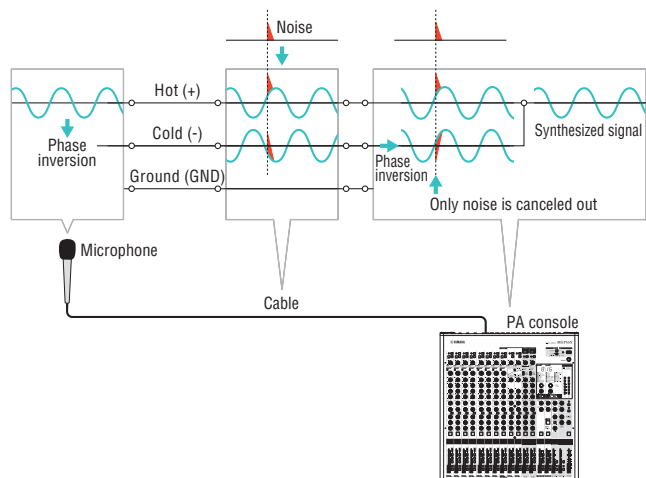
Balanced cables normally have hot and cold insulated conductors inside an outer ground mesh (or foil) that also serves as a noise shield. Unbalanced lines carry a single-phase signal via cables that have two conductors: hot and ground. Unbalanced cables have a single insulated conductor inside an outer ground mesh (or foil) that serves as a noise shield. Both cable types also have an outer layer of insulation.



- Balanced cable: two inner conductors (hot and cold) are surrounded by a shielding mesh or foil conductor
- Unbalanced cable: a single conductor (hot only) is surrounded by a shielding mesh or foil conductor

How balanced lines reject noise

The longer a cable, the more noise it is likely to pick up. Low-level microphone signals are particularly vulnerable. Noise picked up by a microphone cable is amplified by the mixer and other devices, degrading the quality of the signal and therefore the quality of the final sound. In addition to keeping cables as short as possible to reduce noise, balanced lines can be used to effectively cancel any noise that is picked up.



Noise picked up by cables is a single-phase signal that can be cancelled out by adding an equal signal of the opposite phase to it.

The output from keyboards and playback devices normally has a high signal-to-noise ratio to begin with, so noise is less noticeable even when unbalanced lines are used. But if long cables must be used it is usually a good idea to use a D.I. Box that will convert unbalanced input to a balanced signal that will reject noise over long cable runs.

Note: Avoid using a "splitter cable" or similar to separate the stereo output from keyboards or other instruments into mono channels. Depending on how the split signals are processed, this has the potential to cause phase problems that can reduce the level of or completely cancel out the signal.

"Mic" and "Line" cables

Although they can be used for balanced microphone or line-level signals, balanced cables that have a male XLR connector and one end and a female XLR on the other are often simply referred to as "mic" cables because they are commonly used to connect microphones to mixers. Unbalanced cables with phone connectors at each end are often referred to as "line" cables.

D.I. Box

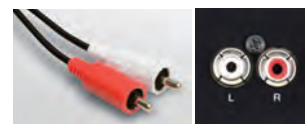
A D.I. Box (Direct Injection Box), converts high-impedance unbalanced signals to low-impedance balanced signals that are more resistant to noise and degradation over long cable runs.

Some D.I. Box are passive and require no power supply, while others are active and must receive power from a battery or the mixer's phantom power supply.

Other connectors used in live sound systems

● RCA pin connectors

These connectors are generally used with home audio and video equipment, and are solely for use with unbalanced signals.



The connectors are color coded: the white connector for the left (L) channel, and the red connector for the right (R) channel.

● USB connectors

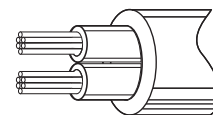
USB connectors are used to transfer audio signals between a USB audio interface equipped mixer and personal computer.



Such connections are useful for recording live performance mixes to a computer, or for music production.

Passive speaker cables

Appropriate speaker cable must be used when connecting a power amplifier to a passive speaker. Cables designed for use with passive speakers are durable and capable of transmitting high-power electrical signals from power amplifiers to speakers. Unlike balanced or unbalanced microphone and line cables, speaker cables are not shielded.



Note: Use microphone or line cables when connecting to powered speakers.

Passive speaker connectors

● SpeakON

This is a dedicated locking connector for passive speaker connections that is used primarily on professional equipment. After inserting the cable, turn it clockwise to lock it.



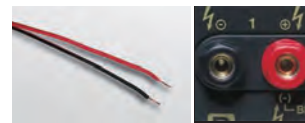
● Phone (for speakers)

Standard phone-connector "line" cables have a single shielded conductor, but dedicated speaker cables are available with two unshielded conductors and phone plug connectors at one end or both ends.



● Zip cord

Zip cord has two conductors joined by an insulating jacket. Zip cord speaker cable is commonly used in home audio systems, and is usually color coded or marked so that is easy to identify the positive and negative conductors. Be sure to connect the power amplifier and passive speakers with the correct polarity: positive to positive, and negative to negative.



Mixer

Mixers are used to adjust the signal levels from multiple inputs to create an ideal balance or “mix.” They can also be used to adjust the tone and other qualities of individual signals as well as the entire mix. Mixers are available in many different configurations with different capacities and features.

Mixer types

Mixers can be broadly classified into three types, as follows:

1: Analog mixers

Analog mixers employ traditional analog circuitry to adjust the volume and other characteristics of input audio signals. Control knobs and faders are directly accessible via the top panel, so it is easy to see current settings and comprehend the overall signal flow. The straightforward layout of analog mixers makes them intuitive to operate.



2: Digital mixers

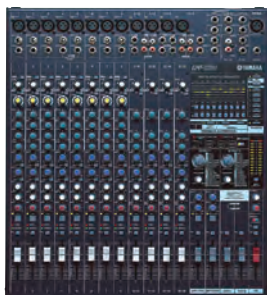
Digital mixers employ digital signal processing technology to process input audio signals, providing a range of control and flexibility that is simply not possible with analog mixers. Digital mixers can store the positions of faders and knobs, and recall them instantly when needed.



Faders and knobs can perform multiple functions, so digital mixers can be made compact, even when they have high channel capacities. Although modern digital mixers are designed to be easy to operate, their complexity and flexibility sometimes mean that a little more study and experience is required for effective setup and operation.

3: Powered mixers

Powered mixers have built-in power amplifiers and can be connected directly to speakers to make up a simple, space-saving sound system. In cases where the equipment is always set up and connected, a powered mixer can simply be switched on and the system is ready for use. The mixer section of a powered mixer can be analog or digital.



Reference:

All-in-one sound systems

All-in-one sound systems are complete packages that include a powered mixer, speakers, and speaker cables. They are easy to configure, easy to carry due to their light and compact format, and well suited to small events and band lineups.



Input channels

The number of input channels provided by a mixer is a vital specification that indicates the number of microphones and musical instruments that the mixer can handle. In addition to the total number of input channels available, it is also important to know how many of those input channels are for microphones, whether line input channels are mono only, and whether stereo inputs are provided.

A band situation might require eight or more microphone channels just to cover the whole drum kit. In such cases it is important to choose a mixer that is equipped with enough microphone channels, and a few extra just in case.

Stereo input channels are important when you need to connect multiple devices that output stereo signals, such as audio playback devices and some keyboard instruments.

● Microphone input channels

Microphones output a very weak signal that must be amplified by a preamplifier, sometimes called a “head amplifier,” in the mixer. Connect microphone cables to the mixer’s MIC connectors.

Note: Phantom power (often labelled as “+48V”) is usually required for condenser microphones.

● Line input channels

Line-level devices such as keyboards and CD players should be connected to the mixer’s LINE connectors. Phone connectors and RCA pin connectors are sometimes used in this case. When a channel provides both MIC and LINE inputs, use the LINE input connector. When a single connector is provided for both MIC and LINE input, reduce the input level by engaging the PAD switch to prevent possible overload and distortion. Generally, phone connectors and RCA pin connectors are used in this case.



Basic Mixer Functions

1: Equalizers

Mixers are often equipped with channel equalizers that can be used to adjust the frequency response (tone) of each channel. Some equalizers have just 2-bands, low and high, while others have three bands: low, mid, and high. 3-band equalizers sometimes provide sweepable midrange control, allowing the frequency of the mid control to be adjusted to target specific instruments and voices. More frequency bands provide more detailed control over the sound.



2: HPF (High Pass Filter)

Microphone and mic/line inputs usually have an HPF function, but some line-only inputs do not. The HPF can be used to cut unnecessary low frequencies right at the channel input, and are frequently used on hi-hat, snare, and vocal channels to create a cleaner sound. An HPF can also minimize annoying popping and breath noise from vocal or speech microphones.



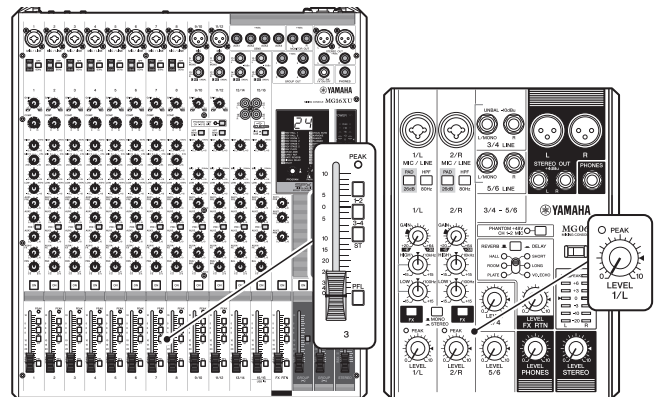
3: PAN

This control adjusts the left-to-right position of the corresponding channel's signal in the output stereo sound field. In live sound applications panning is often used to position each input to correspond to their actual locations on stage. Stereo channels are used for stereo inputs in which the stereo image is predetermined, so a BAL control that allows some overall left-right balance adjustment is provided in place of the PAN control.



4: Level Faders / Knobs

These adjust the level of the corresponding channel, group, stereo output, etc. Linear faders allow for quick operation with easy visual level confirmation. Some mixers have rotary knob type volume controls.



Mixer Outputs

Mixers usually provide multiple outputs for different purposes. The main outputs might feed the main speaker system for the audience, while separate outputs feed stage monitors for the performers. STEREO OUT is commonly used to feed the main speakers, and AUX SENDs are used for stage monitors and external devices. MONITOR OUT is sometimes provided for monitor speakers used in a studio situation. GROUP OUT for grouped outputs, REC OUT for connection to recording devices, and PHONES provides a handy place to connect monitor headphones.

1: AUX bus

The AUX bus is a signal routing function that sends signals to external devices via the AUX outputs. It can be used to feed monitor speakers for the performers separately from the main output, or to send signals to external effects and recording devices. A mixer with multiple AUX outputs should be chosen if separate monitor mixes need to be sent to a number of performers.

2: GROUP bus

The GROUP bus is signal routing function that groups specified channels together for convenient control. For example, if eight microphones channels are used for drums and you want to raise or lower the volume of the entire set, it would be difficult to accurately raise or lower all eight faders. If the same eight channels are assigned to a single group, the volume of the entire drum set can be raised or lowered while maintaining consistent balance between them simply by raising or lowering the group master fader.

3: STEREO bus

The STEREO bus is a signal routing function that routes the mixer's channel, GROUP bus, and other input signals to the STEREO outputs, allowing overall level adjustment via the stereo master fader(s).

Power Amplifier

Power amplifiers amplify line-level signals from the mixer to a level that can drive the system's speakers. Various types of power amplifiers are available, varying widely in output power and functions. Appropriate power amplifiers must be chosen for each live sound situation.

Three major types of power amplifier are described below:

1: Component power amplifiers

Component (or "separate") power amplifiers are solely dedicated to power amplification, but they might also include related functionality, such as speaker processing. Basically, component power amplifiers amplify signals received from the mixer to levels that can drive the system speakers. You might only need one power amplifier, or several installed in one or more racks for a large-scale live sound system.



2: Powered mixers

This is a type of mixer that includes a built-in power amplifier. Since the mixer and power amplifier are integrated into one unit, no external connections between mixer and power amplifier are needed and only one power supply connection is required, significantly simplifying setup and operation.



3: Powered speakers

This type of speaker includes a built-in power amplifier. Since the built-in power amplifier is designed specifically for the speaker, amp and speaker matching can be optimized to deliver the best possible performance. The mixer's line-level outputs can connect directly to powered speakers, so fewer cables are required.



Note: Speakers that do not have a built-in power amplifier are known as "passive speakers".

Tip: For the simplest possible sound system with the minimum number of cables and connections, use either a powered mixer or powered speakers. If your needs are more complex or likely to grow, however, component power amplifiers offer greater freedom for equipment placement and future expansion.

Choosing power amplifiers

In this section we'll cover some points that need to be considered when choosing a power amplifier for passive speakers.

The output requirements for the power amplifier are determined by the number of speakers to be connected and their impedance. To avoid speaker damage, amplifier output power should not exceed the power handling capacity of the speakers.

Amplifier and speaker matching

The Yamaha PX3 power amplifier and Yamaha CBR10 passive speakers will serve as a good example of amplifier and speaker matching. The PX3 spec sheet lists the amplifier's output power and measurement conditions. Looking at a speaker spec sheet, check the power rating PGM value and impedance.

PX3 power amplifier specifications

Output Power: 1kHz Non-clip 20msec Burst Both channels driven	4ohms	500W x 2
	8ohms	300W x 2
	8ohms: Bridge (Power Boost Mode)	600W

CBR10 passive speaker specifications

General Specifications		CBR10
Power Rating	Noise	175W
	PGM	350W
	PEAK	700W
Nominal Impedance		8Ω

Since the PX3 power amplifier can produce 300W output when driving an 8-ohm speaker, and the CBR10 speaker has a rated power capacity (PGM) of 350W and an 8-ohm nominal impedance, the PX3 and CBR10 are a good match.

Tip: The speaker's rated power handling capacity and power amplifier output don't have to be precisely matched. It is safe to select a power amplifier output value of around 0.8 to 1.25 times the speaker's rated PGM power handling capacity. There are a variety of possible amplifier and speaker connection configurations that have an impact on impedance and available power. Continue reading for further advice.

Speaker Impedance

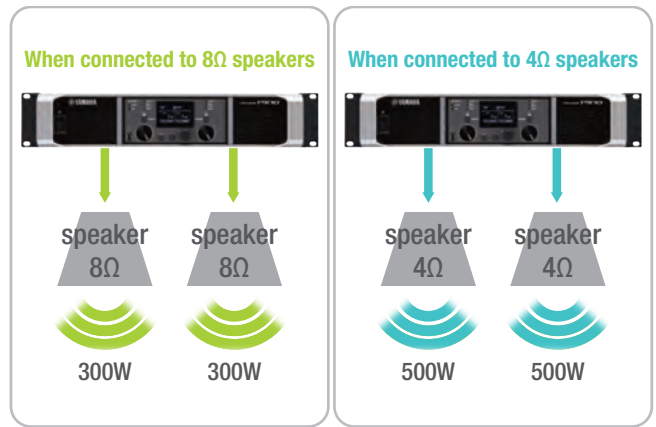
Impedance is basically resistance to AC signals (audio signals are a form of AC, or Alternating Current), measured in ohms (Ω). Higher impedance makes it harder for AC signals to flow, while lower the impedance allows AC signals to flow more easily. With an amplifier of a given power output, the lower the impedance of the speaker connected, the higher the power that can be delivered by the amplifier.

The PX3 can deliver a maximum of 300 watts per channel when it is connected to 8-ohm speakers, but it can put out as much as 500 watts per channel when connected to 4-ohm speakers.

PX3 power amplifier specifications

Output Power: 1kHz Non-clip 20msec Burst Both channels driven	4ohms	500W x 2
	8ohms	300W x 2
	8ohms: Bridge (Power Boost Mode)	600W

Note: If additional speakers are connected them in parallel, the total speaker impedance will change, resulting in a different power output. See "Impedance and Power with Different Speaker Connections" on page 51.



Bridged Connection

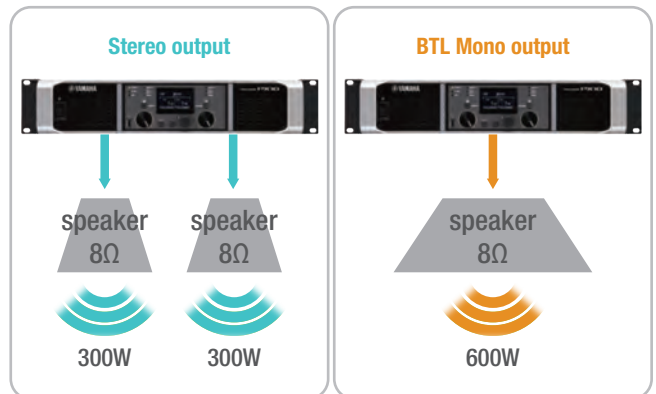
Power Boost mode on the PX3 and PX5 and bridged connections
Stereo power amplifiers can sometimes be connected in bridged (BTL: Bridge-Tied Load, also known as Balanced Transformer Less) mode and used as a single-channel mono amplifier with higher power output.

The PX3 will deliver 300W + 300W output when connected to two 8-ohm speakers in normal stereo mode. When bridged in the Power Boost mode, however, the mono PGM output becomes 600W into an 8-ohm speaker. Stereo power amplifiers are often used in bridged mode to power large sub-woofers.

PX3 power amplifier specifications

Output Power: 1kHz Non-clip 20msec Burst Both channels driven	4ohms	500W x 2
	8ohms	300W x 2
	8ohms: Bridge (Power Boost Mode)	600W

Note: The PX3 can be set to the Power Boost mode by selecting "FULL (MONO)" in the SP TYPE (speaker type) page on the front-panel display. Refer to the PX3 manual for details.



Speakers

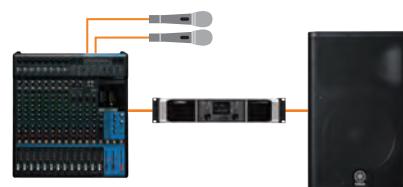
Speakers convert electrical signals from the amplifier into air vibrations that can be sensed by the human ear as sound. Speakers are available in various sizes and configurations for different applications.

Passive or Powered

In terms of how they are driven, speakers fall into two main categories: passive and powered (or “active”).

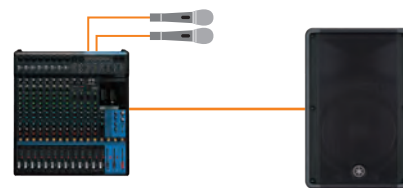
1: Passive speakers

Speakers that require an external power amplifier to produce sound are “passive” speakers.



2: Powered speakers

Speakers that have an internal power amplifier are known as “powered” or “active” speakers. Because active speakers have a built-in power amplifier, they can be directly connected to the line-level outputs of a mixer, thus reducing the system’s overall component and cable count. Also, because the built-in power amplifier is designed specifically for the speaker, matching can be optimized to achieve the highest possible performance.



Yamaha DXR and DBR series active speakers allow for direct microphone input and provide basic mixer functionality built right into the speaker.



Tips: For the simplest possible sound system with the minimum number of cables and connections, use powered speakers. If your needs are more complex or likely to grow, however, passive speakers and separate power amplifiers offer greater freedom for equipment placement and future expansion.

Speakers for different purposes

Various types of speaker are used in live sound systems, each with a different purpose.

1: Main speakers

Sometimes referred to as FOH (Front of House) speakers, these are the speakers that deliver most of the sound to the audience. The size and number of speakers required will depend on the size of the venue, as well as the amplification to be used in the case of passive speakers.



2: Subwoofers

Subwoofers are designed specifically for low-frequency playback, and are used in combination with the main speakers to reinforce the bass frequencies.



3: Monitor speakers

Monitor speakers are provided for the performers, allowing them to hear their own sound as well as that of the other performers. Monitor speakers can be positioned in various ways, depending on their shape and mounting facilities.



Speaker Size, Impedance, Power Rating, & SPL

Specifications that should be checked when choosing passive speakers include speaker (cone) size, impedance, power rating, and sound pressure level (SPL).

Look for the speaker size of the LF speaker unit in the “Components” or “Speaker Unit” specification category. In general, the larger the speaker diameter, the more efficient it will be at reproducing low frequencies.

General Specifications		CBR10	CBR12	CBR15
Components	LF	10" Cone, 2" Voice Coil (Ferrite Magnet)	12" Cone, 2" Voice Coil (Ferrite Magnet)	15" Cone, 2.5" Voice Coil (Ferrite Magnet)
	HF	1" Voice Coil, Compression Driver (Ferrite Magnet)	1.4" Voice Coil, Compression Driver (Ferrite Magnet)	1.4" Voice Coil, Compression Driver (Ferrite Magnet)

Impedance is also a vital specification. In the chart below, the impedance for all speakers is 8Ω (ohms).

General Specifications	CBR10	CBR12	CBR15
Nominal impedance	8Ω	8Ω	8Ω

Check the maximum input (power rating/capacity). The power rating shows the amount of power, in watts, the speaker is designed to handle. A number of power rating values may be listed, each corresponding to different measurement conditions. In general, refer to the PGM (“Music” in some cases) power rating.

General Specifications		CBR10	CBR12	CBR15
Power Rating	NOISE	175W	175W	250W
	PGM	350W	350W	500W
	PEAK	700W	700W	1000W

Lastly, check the speaker's maximum output sound pressure level (SPL). Like the power rating, there may be multiple SPL values corresponding to different measurement conditions. The peak (max) value is the maximum output SPL level.

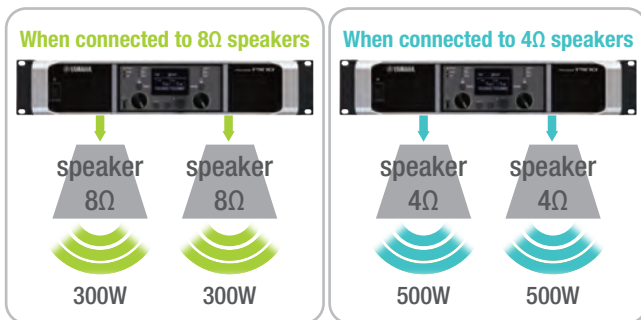
General Specifications		CBR10	CBR12	CBR15
SPL	Sensitivity (1W; 1m on axis)	94dB SPL	96dB SPL	96dB SPL
	Peak (Calculated)	123dB SPL	125dB SPL	126dB SPL

Once the speakers have been selected, choose a power amplifier that matches the speaker specifications. See the "Impedance and Power with Different Speaker Connections" section below for more details.

Note: When choosing a powered speaker, you will need to check the specifications of the built-in power amplifier as well as the speaker specs described above.

Impedance and Power with Different Speaker Connections

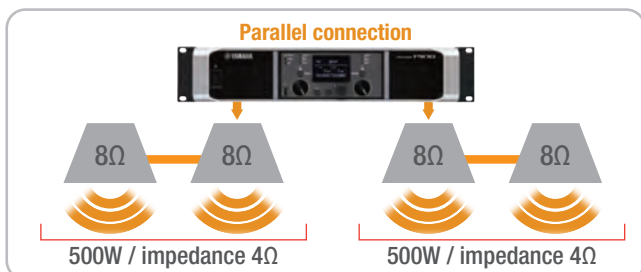
The amount of power a power amplifier can deliver will change according to the impedance of the connected speakers. For example, the output of the Yamaha PX3 power amplifier is different when connected to a pair of 8-ohm speakers versus a pair of 4-ohm speakers as shown below. Although the amplifier doesn't change, output power will be 300W + 300W when connected to a stereo pair of 8-ohm speakers, but will increase to 500W + 500W when connected to a stereo pair of 4-ohm speakers.



PX3 power amplifier specifications

Output Power: 1kHz Non-clip 20msec Burst Both channels driven	4ohms	500W x 2
	8ohms	300W x 2
	8ohms: Bridge (Power Boost Mode)	600W

More speakers can be added by connecting them in parallel, but that changes the total impedance and results in a corresponding change in power amplifier output. Impedance is basically resistance to AC signals. Higher impedance makes it harder for AC signals to flow, while lower the impedance allows AC signals to flow more easily. The lower the impedance, the higher the power delivered by the power amplifier. When two 8-ohm speakers are connected in parallel, their total impedance becomes 4 ohms. The PX3 power amplifier is capable of driving 4-ohm speakers, and its output power will increase in accordance with the reduced impedance.



PX3 power amplifier specifications

Output Power: 1kHz Non-clip 20msec Burst Both channels driven	4ohms	500W x 2
	8ohms	300W x 2
	8ohms: Bridge (Power Boost Mode)	600W

The benefits to connecting speakers in parallel include increased output, as well as the ability to angle individual speakers to cover a wide area. Speakers are often added at live venues to enhance the audio system power and coverage.

With only one speaker per side



With two speakers per side



[Caution]

Although there are benefits to adding speakers as describe above, the total impedance must be kept within a range that is safe for use with the power amplifier.

For example, when two 4-ohm speakers are connected in parallel the total impedance becomes is 2 ohms. Care is required when driving speakers at low impedances like this, because greater load is placed on the power amplifier. The PX3 does not support 2-ohm connections, and cannot be used safely with speaker systems that have a total impedance of 2 ohms.

Calculating total impedance for parallel connections

R1 (impedance of speaker #1) = 4 ohms
R2 (impedance of speaker #2) = 4 ohms
R (total impedance) = 2 ohms

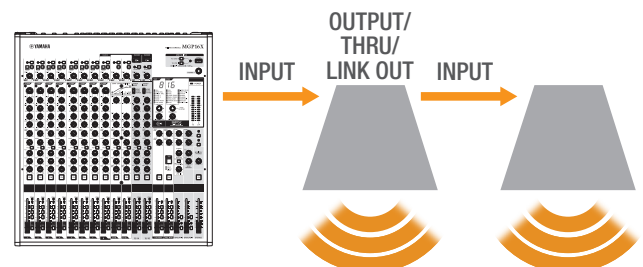
$$\frac{1}{R} = \frac{1}{R_1} + \frac{1}{R_2}$$

A good understanding of the relationship between amplifier output and impedance is essential in building an efficient and safe live sound system.

Tip: Passive speakers like the Yamaha CBR12 have dual connectors that allow convenient parallel connection. The signal from the power amplifier is received at one of the two input connectors (phone or SpeakON connector) via a compatible speaker cable, and the remaining connector is connected to the second speaker via another dedicated speaker cable.

Adding powered speakers

Careful attention must be paid to total impedance when parallel-connecting passive speakers. Powered speakers allow much easier expansion that doesn't involve impedance limitations. Yamaha DSR, DXR, and DBR series powered speakers have connectors labeled OUTPUT/THRU/LINK OUT that can be directly connected to the input of other powered speakers via line cables with XLR connectors.



Reverb

Reverb simulates the natural reflections that occur in acoustic spaces (reverberation), adding a sense of spatial depth and dimension to sound.

Reflections in a room

Reverb is the sum result of sound reflecting off walls, floor, ceiling, and other surfaces in an enclosed space. When you make a sound in a room, the first thing you hear is the direct sound itself. The direct sound is followed by a multitude of reflections off surrounding surfaces. The initial and subsequent reflections are collectively referred to as “reverberation.”



Common reverb types

There are many types of reverb, simulating the reflections that occur in a variety of environments. Some of the most common types are:

Hall: As the name suggests, this type of reverb simulates a concert hall with a long reverb time. This is a relatively prominent reverb effect.

Stage: This type of reverb simulates the sound heard by a performer on stage in a reverberant hall.

Room: This variation simulates the ambience of a smaller room, with a shorter reverb time.

Plate: Mechanical plate reverb units were once common in recording studios. This type of reverb effect simulates the plate reverb sound, and is often used to add warmth to vocals.

Hints for using reverb

Although reverb can add beautiful ambience, it can make the sound muddy if overused. Add reverb a little at a time, making sure that you can hear the original sound clearly.

Compressor

A compressor is a signal processor used to control dynamic range (the difference between the loudest and softest levels) by “compressing” the sound.

Threshold, ratio, and gain

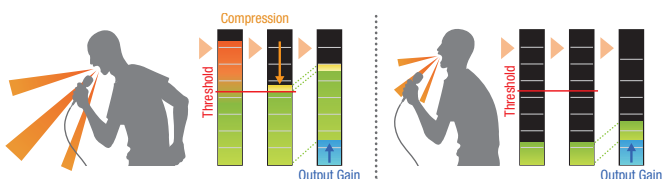
The three primary parameters of a compressor are threshold (the volume at which the compressor begins operating), ratio (the degree of compression), and output gain (gain that compensates for the loss in overall level caused by compression).

The threshold determines the level at which compression will begin. Signals that exceed the threshold are reduced in level by an amount determined by the ratio parameter. Final overall level can be adjusted as necessary using the output gain control. Lowering signal peaks while boosting the lowest signal levels reduces dynamic range and, when applied appropriately, can result in cleaner, smoother mixes that are more comfortable to listen to.

Why use compression?

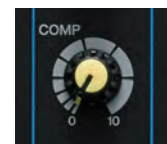
Vocal and instrumental performances rarely maintain a constant level. Keeping levels under control so that they fall within a comfortable range that also contributes to overall clarity is an important element of any mix.

Vocal performances can have a very wide dynamic range. Just the right amount of compression on vocals can prevent overload when the vocalist decides to really belt out a note while simultaneously keeping the overall vocal level in balance with the other instruments. Compression is often used on bass guitar to create a smoother, cleaner sound that fits better in the mix. Snare drum and other instruments with a sharp attack benefit from compression that can fatten the sound and give it more impact. In addition to controlling the sound and mix from a musical perspective, compression can protect equipment from sudden sharp level spikes.



Professional sound with 1-Knob Compressors

Experienced engineers can use compressors to revitalize tracks and add life to guitar parts, punch to bass lines, tightness to snare drums, presence to vocals, and much more. But poorly applied compression can rob the sound of its dynamic appeal.



The Yamaha 1-Knob Compressor has just one knob to replicate the control a seasoned engineer might apply. This simple, single-knob compressor lets you easily achieve ideal compression for vocals and instruments, giving your mixes some professional polish without the need to learn complex techniques, or the need to hook up external compression devices.

A number of Yamaha mixers feature 1-Knob Compressors, including the TF series, MGP series, MG series (excluding the MG06/06X) and EMX7/5. The 1-Knob Compressor is ideal for mixing live music.

Equalizer

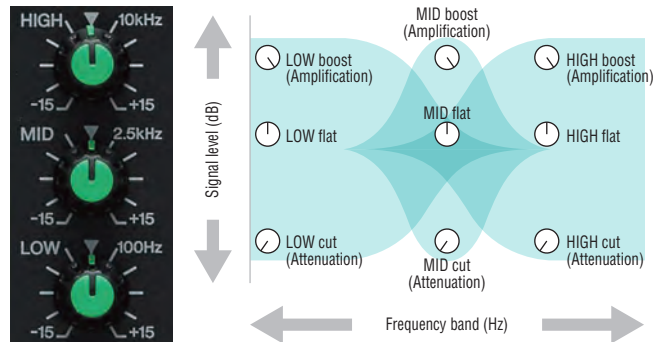
An equalizer can be used to boost or cut specific frequency bands to adjust the character of the sound.

Different types of equalizer are used for different purposes. Two-band, three-band, or even four-band “channel equalizers” are often provided on individual mixer channels for precise response shaping, while graphic equalizers with 7, 15, or 31 controllable bands are sometimes used for overall response shaping or room “tuning.”

Channel equalizers

Channel equalizers usually provide control over several separate frequency bands. When the high-end needs a little boost or the low-end needs to be cut, the corresponding controls can simply be rotated clockwise to boost, or anti-clockwise to cut. This type of intuitive control is a feature of many mixers. Some variations include two-band (high and low) types, three-band (high, mid, and low) types, and four-band (high, high-mid, lo-mid, and low) types. The channel equalizers on some high-performance mixers such as Yamaha TF series digital mixing consoles allow adjustment of the frequency band and bandwidth in addition to basic boost/cut.

A general rule to keep in mind when using equalizers is that it is usually better to cut rather than boost. If boost must be applied it should be applied very gently. Consider the approximate frequency range of each individual instrument or voice, and try cutting any frequencies that fall outside their range. For example, there is very little useful low-frequency content in the sound of a hi-hat cymbal, so we may as well cut the low frequencies using the equalizer controls so that unwanted low frequencies aren't amplified by that channel.



This table shows examples of the approximate fundamental tonal frequency range covered by common live sound sources.

Bass Guitar	30Hz	330Hz
Kick Drum	50Hz	500Hz
Guitar	80Hz	1.3kHz
Snare Drum	100Hz	400Hz
Male Voice	100Hz	900Hz
Female Voice	250Hz	1.2kHz
Piano (88 notes)	27Hz	4.2kHz

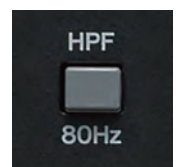
Graphic equalizers

Graphic equalizers are commonly connected to the mixer outputs to adjust the overall response of the sound, or to correct feedback problems. The fixed frequency bands of a graphic equalizer are boosted or cut via linear controls. The photo shows a 31-band stereo-type equalizer.



High-pass filters

High-pass filters (HPF) attenuate frequencies below the frequency they are set to (they “pass” higher frequencies). Although not technically an “equalizer,” a high-pass filter does share characteristics with equalizers in that it cuts a specified range of frequencies. High-pass filters can be useful to reduce vocal popping and breath noise, or eliminating unwanted low-frequency “spill” from a bass drum into a hi-hat microphone, for example. An HPF can also be effective on some string, brass, and woodwind instruments.





Products

Digital Mixers

Analog Mixers

Powered Mixers

Powered Loudspeakers

Loudspeakers

Power Amplifiers

Portable Sound Systems

Studio Monitor Speakers

Mixer with USB Audio Interfaces

Headphones

P roducts

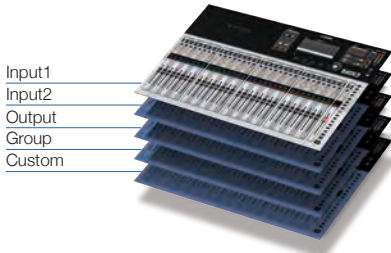
Digital Mixers

Creative Freedom

Yamaha TF series and 01V96i digital mixing consoles support experienced sound engineers as well as newcomers to live sound with extensive functionality and creative control.

Digital Advantages

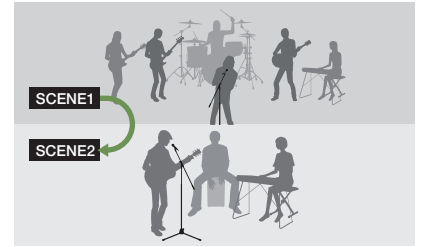
Layers



Digital consoles can have “layers” that allow channel faders and other controls to be easily switched between multiple channels and functions.



Fast, accurate motor-driven faders respond immediately, moving precisely to the settings of the selected mode.



You can instantly switch between complete console setups or “scenes” during a performance for dramatically enhanced efficiency and workflow.



TF Series Smooth, Natural Control Flow

TouchFlow Operation™ introduced in the TF series consoles allows the user to respond to the music and artists on stage with unprecedented speed and freedom, taking live sound reinforcement to a new level of refinement. With the TouchFlow Operation interface optimized for touch panel control, experienced engineers as well as newcomers to the field will find it easier than ever to achieve the ideal mix.

Touch Operation for Intimate Control

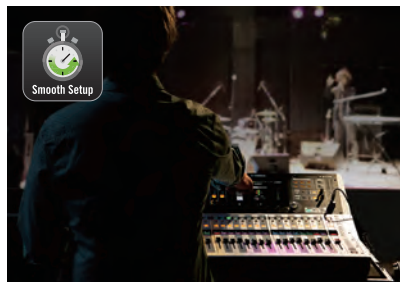
Refined for the smoothest possible operation via touch-panel control, the TF user interface offers a smooth workflow that can be an advantage in any mixing situation. The display content has been specifically designed for easy, direct accessibility, with a layout that promotes the most natural, efficient mixing. Touch panel operation is as easy as shaping the sound with your fingertips.

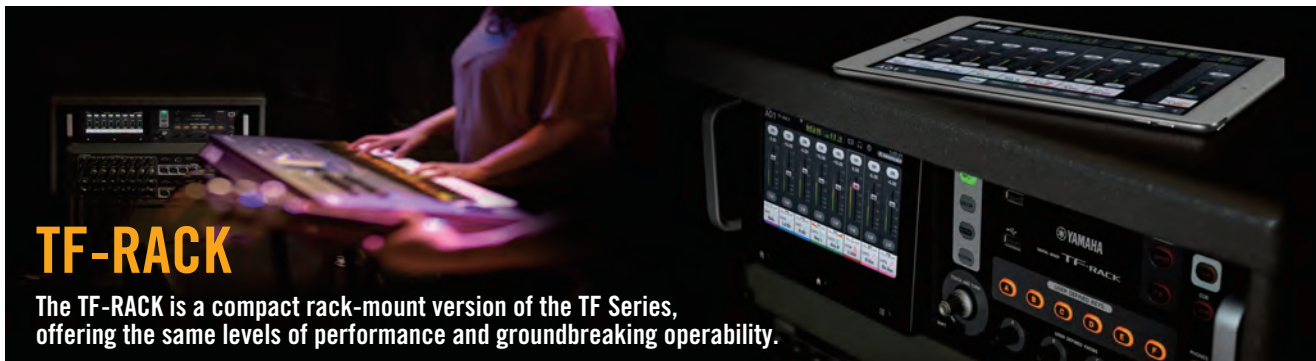
New Features for Smooth Setup and Operation

Effective gain, compressor, and EQ setup requires a considerable degree of skill and experience. Even veteran engineers need to spend time on these important basic settings. In addition to their intuitive user interfaces, TF consoles include a number of features that contribute to faster, smoother setup of these essential parameters.

Applications that Work Seamlessly with the Console

Separate software applications that allow wireless mixing, personal monitor mixing, or offline setup of digital live mixing consoles have become fairly common. The TF series goes a step further with three dedicated apps – TF Editor, TF StageMix, and MonitorMix – that feature user interfaces that seamlessly extend the console’s own on any device.





TF-RACK

The TF-RACK is a compact rack-mount version of the TF Series, offering the same levels of performance and groundbreaking operability.



01V96i

Perfect Capture

The 01V96i will take every aspect of your mixing and recording to the next level, along with 16-in/16-out USB 2.0 audio streaming at an impressive 96kHz, Steinberg's Cubase AI, a full selection of acclaimed Yamaha Virtual Circuitry Modeling DSP effects, and studio-quality preamplifiers.

24bit/96kHz Performance

The 01V96i features top-quality 24bit/96kHz converters for all onboard A/D and D/A conversion, plus 32bit/96kHz internal processing (58-bit accumulator). It also offers a comprehensive range of 96kHz compatible internal stereo effects, independent channel compression and gating/ducking processors, 4-band parametric channel equalizers, plus channel delay.






Powerful Mixing Capacity, Versatility

The 01V96i has tremendous mixing capacity for a console its size, making it extremely versatile and suitable for a wide range of studio and sound reinforcement applications. With up to 40 simultaneous inputs and 20 buses at 96kHz, the 01V96i delivers large console power in a rack-mount sized package. Offering a main stereo bus, eight individual mixing buses, two solo buses, and eight auxiliary buses—a total of 20 in all—the 01V96i gives you plenty of signal-routing options to adapt to just about any mixing requirements.

Full Suite of Yamaha VCM Effects and High-Resolution REV-X Reverbs

Yamaha's acclaimed VCM effects faithfully model the analog circuitry of classic effect units right down to the very last resistor and capacitor, bringing the richness and warmth of analog to a digital environment. The latest "REV-X" algorithms feature the richest reverberation and smoothest decay available, based on years of dedicated research and development by Yamaha.

Specifications

Model					
	TF5	TF3	TF1	01V96i	TF-RACK
Line Input	48 (40 mono + 2 stereo + 2 return)	48 (40 mono + 2 stereo + 2 return)	40 (32 mono + 2 stereo + 2 return)	40 (32 Mono + 4 Stereo)	40 (32 mono + 2 stereo + 2 return)
Mic Input	32	24	16	12	16
Aux	20 (8 mono + 6 stereo)	20 (8 mono + 6 stereo)	20 (8 mono + 6 stereo)	8	20 (8 mono + 6 stereo)
DCA Group / Group	8 DCA Groups	8 DCA Groups	8 DCA Groups	8 Groups	8 Groups
Fader Configuration	32 + 1 (Master)	24 + 1 (Master)	16 + 1 (Master)	16 + 1 (Master)	—
TouchFlow Operation™	Yes	Yes	Yes	—	—
Gain Finder™	Yes	Yes	Yes	—	Yes
Internal Effect	8 Effects + 10 GEQ	8 Effects + 10 GEQ	8 Effects + 10 GEQ	4 Effects	—
APP	TF Editor, TF StageMix, Monitor Mix			01V96 Editor	TF Editor, TF StageMix, Monitor Mix
Included Software	Nuendo Live			Cubase AI	Nuendo Live
Dimensions (W x H x D)	866 x 225 x 599mm (34.1" x 8.9" x 23.6")	716 x 225 x 599mm (28.2" x 8.9" x 23.6")	510 x 225 x 599mm (20.1" x 8.9" x 23.6")	436 x 148 x 548mm (17.1" x 5.9" x 21.6")	480 mm x 132 mm x 409 mm (18 7/8 in x 5 1/4 in x 16 1/8 in)
Net Weight	20.0kg (44.1lb)	17.0kg (37.5lb)	13.5kg (29.8lb)	14.0kg (30.9lb)	9.2kg (20.3 lb)

Analog Mixers

Fill Your Space with Premium Analog Sound

MGP and MG series mixing consoles feature studio-quality preamps, powerful digital processing, and rugged construction. These fine analog consoles easily adapt to an impressive range of applications, from live performances, parties, and corporate events to fixed installations.



MGP Series
Digital Heart. Analog Soul.

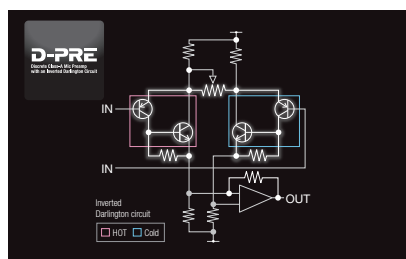


MG Series (XU Model / Standard Model)
Road Tough. Studio Ready.

Model	MGP Series	MG Series (XU Model)	MG Series (Standard Model)
D-PRE (Discrete Class-A Mic Preamps)	YES	YES	YES
1-knob COMP	YES	YES	YES (excluding MG06)
X-pressive EQ	YES	— (excluding MG06X)	—
Digital Effects	YES (REV-X + SPX)	YES (SPX)	—
APP	MGP Editor	MG Rec&Play, Cubase AI & Cubasis LE	—
USB Interface	YES	YES (excluding MG06X)	—

Studio-grade Yamaha Discrete Class-A D-PRE Preamps

D-PRE preamplifiers deliver solid, natural sounding bass and smooth, soaring highs with very low distortion. They utilize an inverted Darlington circuit comprising cascaded transistors in a configuration that provides more power with low impedance. D-PRE preamps possess an impressively wide frequency range without overly emphasizing any specific elements of the sound.



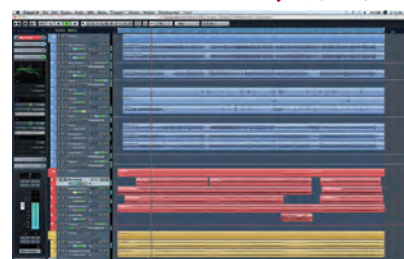
SPX Digital Effects / REV-X Reverb

MGP and MG XU models feature Yamaha's acclaimed SPX digital multi-effect processor, offering a comprehensive suite of editable digital effects that are perfect for enhancing your mix. The MGP series also features high-end REV-X reverb with realistic, high-resolution reverb effects: Hall, Room and Plate. Both processors can be used simultaneously.













CUBASE AI (MG XU models)

MG XU models include authorization for a downloadable version of Steinberg's Cubase AI software. As one of the most advanced, comprehensive DAW suites available, Steinberg's Cubase provides users with a powerful and flexible platform for music production. Cubase AI offers the same core technologies and interface that have won Cubase worldwide acclaim. (MG10XU, MG12XU, MG16XU, and MG20XU)



Specifications

MGP Series				
Model	 MGP32X	 MGP24X	 MGP16X	 MGP12X
Line Input	32 (24 mono + 4 stereo)	24 (16 mono + 4 stereo)	16 (8 mono + 4 stereo)	12 (4 mono + 4 stereo)
Mic Input	24	16	10	6
Aux Send	6 AUX Sends + 2 FX Sends		2 AUX Sends + 2 FX Sends	
Group Out	4 Group Buses + ST Bus			
USB IN/OUT	USB Device, iPod / iPhone		iPod / iPhone only	
Digital Effects	REV-X (8 programs) + SPX (16 programs)			
USB Rec / Play (WAV/MP3)	Yes		—	
Dimensions (W x H x D)	1,027 x 169 x 565mm (40.4" x 6.7" x 22.2")	819 x 169 x 565mm (32.2" x 6.7" x 22.2")	447 x 143 x 495mm (17.6" x 5.6" x 19.5")	348 x 143 x 495mm (13.7" x 5.6" x 19.5")
Net Weight	19.0kg (41.9lb)	15.5kg (34.2lb)	9.0kg (19.8lb)	7.5kg (16.5lb)

MG Series						
Model	 MG20XU	 MG20	 MG16XU	 MG16	 MG12XU	 MG12
Line Input	20 (12 mono + 4 stereo)	20 (12 mono + 4 stereo)	16 (8 mono + 4 stereo)	16 (8 mono + 4 stereo)	12 (4 mono + 4 stereo)	12 (4 mono + 4 stereo)
Mic Input	16	16	10	10	6	6
Aux Send	4 (incl.FX)	4	4 (incl.FX)	4	2 (incl.FX)	2
Group Out	4	4	4	4	2	2
1-Knob Comp	Yes	Yes	Yes	Yes	Yes	Yes
SPX Digital Effects	24 programs	—	24 programs	—	24 programs	—
USB IN/OUT	Yes (2 In / 2 Out)	—	Yes (2 In / 2 Out)	—	Yes (2 In / 2 Out)	—
Cubase AI & Cubasis LE	Yes	—	Yes	—	Yes	—
MG Rec/Play	Yes	—	Yes	—	Yes	—
Dimensions (W x H x D)	444 x 130 x 500mm (17.5" x 5.1" x 19.7")	444 x 130 x 500mm (17.5" x 5.1" x 19.7")	444 x 130 x 500mm (17.5" x 5.1" x 19.7")	444 x 130 x 500mm (17.5" x 5.1" x 19.7")	308 x 118 x 442mm (12.1" x 4.6" x 16.6")	308 x 118 x 442mm (12.1" x 4.6" x 16.6")
Net Weight	7.1kg (15.7lb)	6.9kg (15.2lb)	6.8kg (15.0lb)	6.6kg (14.6lb)	4.2kg (9.3lb)	4.0kg (8.8lb)

MG Series						
Model	 MG12XUK	 MG10XUF	 MG10XU	 MG10	 MG06X	 MG06
Line Input	12 (6 mono + 3 stereo)	10 (4 mono + 3 stereo)	10 (4 mono + 3 stereo)	10 (4 mono + 3 stereo)	6 (2 mono + 2 stereo)	6 (2 mono + 2 stereo)
Mic Input	6	4	4	4	2	2
Aux Send	1 (incl.FX)	1 (incl.FX)	1 (incl.FX)	1	0	0
Group Out	0	0	0	0	0	0
1-Knob Comp	Yes	Yes	Yes	Yes	—	—
SPX Digital Effects	24 programs	24 programs	24 programs	—	6 programs	—
USB IN/OUT	Yes (2 In / 2 Out)	Yes (2 In / 2 Out)	Yes (2 In / 2 Out)	—	—	—
Cubase AI & Cubasis LE	Yes	Yes	Yes	—	—	—
MG Rec/Play	Yes	Yes	Yes	—	—	—
Dimensions (W x H x D)	315 x 91 x 297mm (12.4" x 3.6" x 11.7")	275 x 110 x 390mm (10.8" x 4.3" x 15.4")	244 x 71 x 294mm (9.6" x 2.8" x 11.6")	244 x 71 x 294mm (9.6" x 2.8" x 11.6")	149 x 62 x 202mm (5.9" x 2.4" x 7.9")	149 x 62 x 202mm (5.9" x 2.4" x 7.9")
Net Weight	3.0kg (6.62 lbs)	3.3 kg (7.3 lbs)	2.1kg (4.63lb)	1.9kg (4.19lb)	0.9kg (1.98lb)	0.9kg (1.98lb)

Powered Mixers

Power. Portability. Performance.

Ever since the launch of the first EM series integrated mixer in 1976, and the subsequent release of the first EMX console, Yamaha has remained the industry leader for powered mixers. Each model of the new EMX lineup is an integrated solution that combines a power amplifier with a versatile mixer in a single, compact unit with functionality and professional performance features tailored to the demands of various sound reinforcement environments.

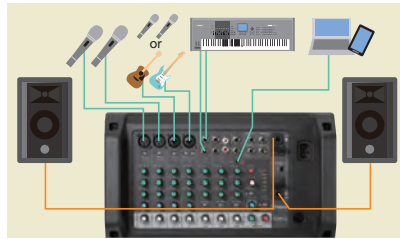
Key Features

Easy Setup, Easy Operation



With their intuitive, user-friendly architecture, the EMX series can be setup and operated quickly and easily

Complete Range of Input & Output Channels



Despite their compact construction, they provide all the tools you need to mix, process, and deliver your audio just the way you want.

Feedback Suppression



Yamaha's unique Feedback Suppressor instantly eliminates annoying feedback. (EMX7/5/2, EMX5016CF)



The EMX7 and EMX5 were designed specifically for musicians, performers, or public speakers who put a premium on sound quality and convenient operation, and are ready to take their sound reinforcement to the next level. Only speakers and a microphone are required to configure a fully functioning, portable, reliable sound system with all the tools you need to mix, process, and deliver your audio exactly as you want it, anywhere you want it.

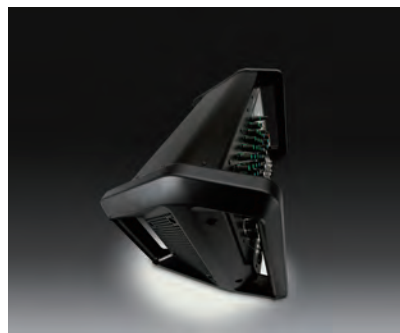
Powerful Amplifiers

An internal Class-D power amplifier delivering 630 watts in the EMX5 or 710 watts in the EMX7 provides high output while consuming surprisingly little power.



Portable and Robust

A unique design allows for flexible layout and incorporates large handles and an impact-resistant metal chassis.



4 Yamaha Reverbs for Additional Atmosphere

HALL mimics the sound characteristics of a large hall, PLATE for a more classic, defined sound, ROOM adds a more intimate sound to performances, and ECHO gives instruments or vocal a more professional edge, as well as being an attractive option for KARAOKE.



EMX5016CF / EMX5014C

Serious Live Sound Capability Plus Innovative Digital Features



The EMX5016CF and EMX5014C are console type powered mixers with intuitive operability. They are remarkably compact and portable for live sound systems with this much capability, offering performance and reliability that will satisfy discerning professional users on the road as well as in installed applications.

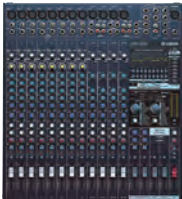

Specifications

Box Type

Model				
		EMX7	EMX5	EMX2
Output Power*	4 Ω	710 W + 710 W	630 W + 630 W	250 W + 250 W
	8 Ω	500 W + 500 W	460 W + 460 W	170 W + 170 W
Line Input		12 (4 mono + 4 stereo)	12 (4 mono + 4 stereo)	10 (4 mono + 3 stereo)
Mic Input		8	8	4
Aux Send		2	2	0
Feedback Suppressor		Yes	Yes	Yes
1-Knob Comp		Yes	Yes	Yes
Digital Effects		SPX (24 programs) x 1	SPX (24 programs) x 1	REVERB (4 programs) x 1
Flex 9 Graphic Equalizer		Yes	Yes	Yes
1-Knob Master EQ™		—	Yes	Yes
Speaker Processor and Preset		Yes	Yes	Yes
Dimensions (W x H x D)		465 x 308 x 325mm (18.3" x 12.1" x 12.8")	465 x 308 x 325mm (18.3" x 12.1" x 12.8")	375 x 147 x 220mm (14.8" x 5.8" x 8.7")
Net Weight		9.5kg (20.9lb)	9.5kg (20.9lb)	4.2kg (9.3lb)

*1 kHz THD+N < 10% (CEA2006)

Console Type

Model		
	EMX5016CF	EMX5014C
Power* (4 Ω)	500 W + 500 W	500 W + 500 W
Line Input	16 (8 mono + 4 stereo)	14 (6 mono + 4 stereo)
Mic Input	12	8
Aux Send	4 (incl. 2x Effect Send)	3 (incl. 1x Effect Send)
Feedback Suppressor	Yes	—
Feedback Channel Locating	—	Yes
1-Knob Comp	Yes	Yes
Digital Effects	SPX (16 programs) x 2	SPX (16 programs) x 1
9-band Digital Graphic EQ	Yes	Yes
FRC (Frequency Response Correction) System	Yes	—
Multi-band "Maximizer"	Yes	—
Dimensions (W x H x D)	444 x 155 x 493mm (17.5" x 6.1" x 19.4")	444 x 155 x 493mm (17.5" x 6.1" x 19.4")
Net Weight	11.0kg (24.3lb)	10.5kg (23.1lb)

*1 kHz THD+N < 0.5%. Output power at 8Ω vary by electricity voltage and regions.

Powered Loudspeakers

Power and Performance. Integrated.

Built around the core concept of delivering the loudest and clearest sound possible even at maximum output, Yamaha powered loudspeakers were designed to give users a significant advantage for applications where they're pushing their speakers to the absolute limit.



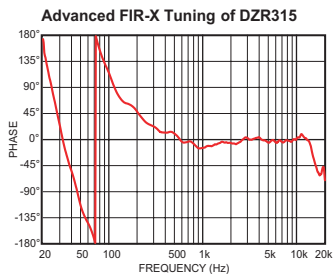
DZR series

State-of-the-Art Powered Performance

Best-in-class SPL performance, sophisticated DSP processing, and unrivaled low-frequency reproduction. DZR series powered loudspeakers deliver on all fronts, realizing truly professional sound delivered with power that belies their compact, lightweight design.

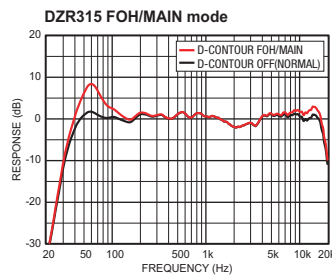
Yamaha DSP: Intelligent Processing for Consistent Clarity and Power

Advanced FIR-X Tuning



Newly developed Advanced FIR-X tuning improves significantly on our proprietary FIR-X tuning technology, achieving very smooth frequency response for stunning clarity and imaging without distortion.

D-CONTOUR



D-CONTOUR is an intelligent multi-band compressor that ensures your DZR sound stays consistent and clear no matter how hard the speakers are driven. D-CONTOUR provides two settings: FOH/MAIN and MONITOR.

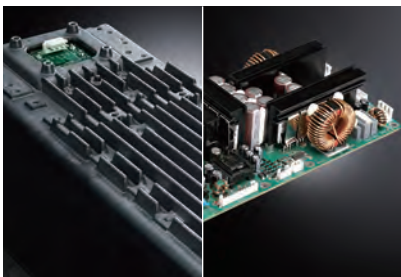
Presets for DZR/DXS XLF Combinations



An onboard LCD screen allows users to easily navigate the variety of DSP functions, load presets, and make precise adjustments to PEQ, delay, and routing. Users can easily transfer saved settings to a USB flash drive.

Key Features

2000-watt Class-D Amplification



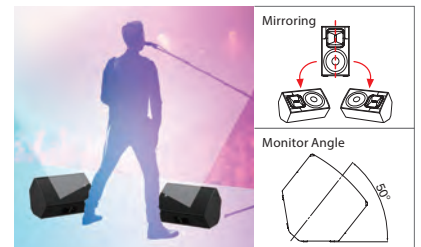
An all-new, highly efficient Class-D amplification module produces best-in-class 2000-watt power and impressive 143 dB SPL.

Premium-Grade Custom-Designed Transducers



We thoroughly analyzed a wide range of transducers, choosing low-distortion units that provide extremely flat frequency response, wide dynamic range and high power handling.

Power in Flexibility



The enclosure design for the DZR15 and DZR12 and their Dante versions enables mirror-mode monitor placement that produces a symmetrical sound field for the performer, creating a large, well-defined "sweet spot." (The DZR10 and DZR10-D are asymmetric.)

DXS XLF series

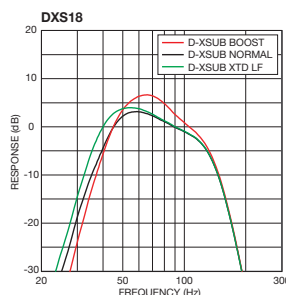
How Low Can You Go?



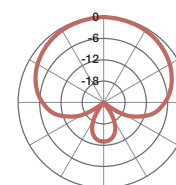
Not for the faint of heart, the newest DXS XLF's go deeper than any model before, with an extended low frequency of 30Hz— perfect for professional touring musicians, DJ's, and sound engineers who need that extra-low, thundering bottom end that you just can't get from other subwoofers in this class.

Powerful DSP for Flexible Control

Onboard D-XSUB LF processing gives you total management of your lower frequencies with NORMAL, BOOST, or XTENDED LF mode. With DXS-XLF setups consisting of two or more subwoofers, selecting the Cardioid Mode effectively decreases the sound pressure on stage while increasing the bottom-end output directed towards the audience.



DXS18 - Cardioid Mode
Horizontal Polar
Pattern @ 50Hz



Advanced Acoustics and Utility

New Yamaha proprietary transducers with a durable die-cast aluminum frame and 4" voice coil are crucial to achieving such impressive low frequency response, even at the highest output levels. A dual damper also helps to realize extraordinarily high linearity of the low frequencies by mitigating unwanted vibration during extreme output levels.



The First Dante-equipped Professional SR Loudspeakers

All DZR speakers and DXS XLF subwoofers come in Dante-equipped "D" models enabling smart system integration with Yamaha consoles and other Dante devices via the Dante I/O (2in/2out) which also features SRC (sample rate conversion) allowing 48kHz device compatibility.

Patching is also simplified considerably with Yamaha CL/QL digital mixing consoles from the I/O device display, while TF console users can select Quick Config for surprisingly fast, easy set up and operation of a sophisticated audio system. Additionally, Yamaha's customizable control panel software "ProVisionaire Control" for Windows and "ProVisionaire Touch" for iOS (iPad) are also Dante compatible, adding simpler remote management for sound reinforcement systems.



Model	Power Rating*1	Maximum SPL *2	Frequency Range (-10dB)	LF	MF	HF	Coverage Angle
DZR315/315-D	2000W	143dB SPL	31Hz - 20kHz	15" Cone, 3" VC	8" Cone	2" VC, 1" Throat	H75° x V50°
DZR15/15-D	2000W	139dB SPL	34Hz - 20kHz	15" Cone, 3" VC	—	2" VC, 1" Throat	H90° x V50°(Rotatable)
DZR12/12-D	2000W	139dB SPL	39Hz - 20kHz	12" Cone, 3" VC	—	2" VC, 1" Throat	H90° x V60°(Rotatable)
DZR10/10-D	2000W	137dB SPL	44Hz - 20kHz	10" Cone, 3" VC	—	2" VC, 1" Throat	H90° x V60°(Rotatable)
DXS18XLF/18XLF-D	1600W	136dB SPL	30Hz - 150Hz	18" Cone, 4" VC	—	—	—
DXS15XLF/15XLF-D	1600W	136dB SPL	33Hz - 150Hz	15" Cone, 4" VC	—	—	—

*1 Dynamic power *2 Measured Maximum SPL (peak)

Powered Loudspeakers

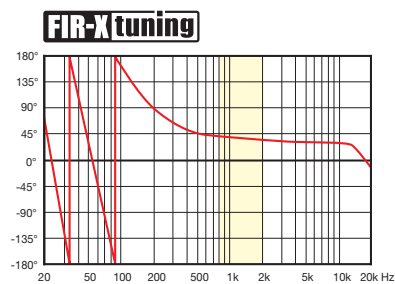


**DXR series,
DBR series**
POWER TO THE PEOPLE

Performance runs in the family, DXR Series and DBR series loudspeakers offer superb high-definition sound and class-leading sound pressure levels—yet in a more compact and extremely functional design that provides a level of versatility unmatched by any speaker in its class today

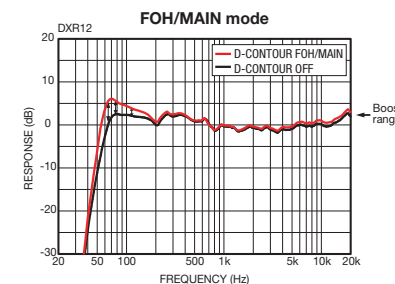
The Power of Digital

Ultra-Precise DSP Processing for High Definition Sound



FIR-X™ tuning simultaneously optimizes frequency and phase response while adjusting the time alignment between the HF and LF transducers. This creates a very smooth response around the crossover point, providing much better clarity and imaging than what is possible with typical crossovers.

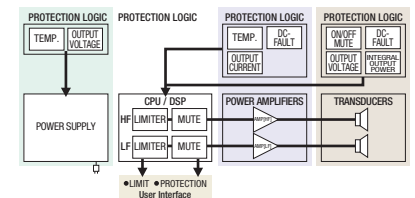
Intelligent Dynamic Control for Consistent Clarity at Any Output Level



D-CONTOUR is an intelligent multi-band compressor that ensures powerful, consistent sound at any output level. D-CONTOUR provides two settings: FOH/MAIN and MONITOR.

Extensive DSP Protection Functions for Maximum Output

One of the advantages of active loudspeakers is that the combination of transducers and amplifiers can be perfectly matched and optimized for the best possible performance. During the development of the DSR, DXR and DXS series, we measured and tested the durability of each transducer and the overall amp output through countless indoor and outdoor listening tests. Based on the test results, we were able to set the optimal limiter point for each model using precise DSP control.



Power to Play

Best-in-class Power



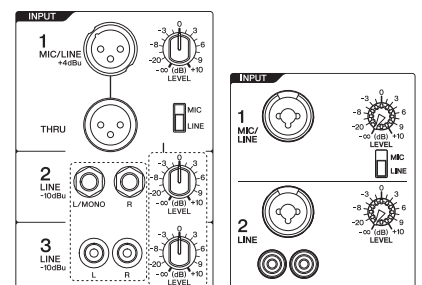
High-performance Yamaha Class-D amplifiers are capable of producing up to an impressive 1100 watts of power with a class-leading SPL of 133 dB. Combined with meticulously selected, custom-designed transducers, Yamaha powered loudspeakers deliver powerful, well-defined sound with exceptional clarity and accuracy as well as minimal distortion.

Flexible set up



Built tough to endure the rigors of the road, the sturdy, durable enclosures also come equipped with flexible features that allow each model to seamlessly adapt to a wide variety of sound reinforcement environments and applications.

Built-in Mixer



DXR series

DBR series

The DXR and DBR Series feature a flexible onboard mixer with a variety of inputs that is ideal for simple vocal/ instrument amplification.

DXS series

Serious Bass Performance

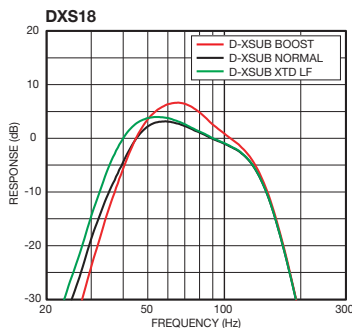
DXS Series were made for audio engineers and bass fanatics who demand consistent high-quality sound for their live performances or DJ events- even when it's cranked to the maximum SPL.

Band-Pass Type Enclosure



All DXS Series enclosures incorporate a band-pass design to produce extremely high SPL by reducing unwanted vibrations and noise that could otherwise sabotage your sound.

D-XSUB Bass Processing



Powerful D-XSUB processing on the DXS series gives you total control of your low frequencies.

Durable, Portable, Road Ready



Enclosure exteriors feature a durable, damage-resistant polyurea coating to protect the cabinet. Optional casters are available for faster, more convenient positioning and setup.

Specifications

Model	DXR15	DXR12	DXR10	DXR8	
System Type	2-way, Bi-amp powered speaker, Bass-reflex type				
Components	LF	15" cone, 2.5" voice coil, Ferrite Magnet	12" cone, 2.5" voice coil, Ferrite Magnet	10" cone, 2" voice coil, Ferrite Magnet	8" cone, 2" voice coil, Ferrite Magnet
	HF	1.4" diaphragm, 1" throat compression driver, Ferrite Magnet			
Frequency Range (-10dB)	49Hz - 20kHz	52Hz - 20kHz	56Hz - 20kHz	57Hz - 20kHz	
Output Power*1 (Dynamic)	1100W				
Maximum SPL (1m, on axis)	133dB SPL	132dB SPL	131dB SPL	129dB SPL	
Floor Monitor Angle	50° Symmetrical		50°	—	
Material, Finish	ABS, Matte black				
Dimensions (W x H x D)	445 x 700 x 380 mm (17.5" x 27.5" x 15.0")	362 x 601 x 350 mm (14.2" x 23.6" x 13.7")	305 x 502 x 310 mm (12.0" x 19.7" x 12.2")	280 x 458 x 280 mm (11.0" x 18.0" x 11.0")	
Net Weight	22.5kg (49.6lbs)	19.3kg (42.5lbs)	14.6kg (32.2lbs)	13.5kg (29.8lbs)	

Model	DBR15	DBR12	DBR10	
System Type	2-way, Bi-amp powered speaker, Bass-reflex type			
Components	LF	15" cone, 2.5" voice coil, Ferrite Magnet	12" cone, 2" voice coil, Ferrite Magnet	10" cone, 2" voice coil, Ferrite Magnet
	HF	1.4" diaphragm, 1" throat compression driver		
Frequency Range (-10dB)	50Hz - 20kHz	52Hz - 20kHz	55Hz - 20kHz	
Output Power*1 (Dynamic)	1000 W			
Maximum SPL (1m, on axis)	132dB SPL	131dB SPL	129dB SPL	
Floor Monitor Angle	50° Symmetrical		50°	
Material, Finish	Plastic, Black			
Dimensions (W x H x D)	455 x 700 x 378 mm (17.9" x 27.6" x 14.9")	376 x 601 x 348 mm (14.8" x 23.7" x 13.7")	308 x 493 x 289 mm (12.1" x 19.4" x 11.4")	
Net Weight	19.3 kg (42.6 lbs)	15.8 kg (34.8 lbs)	10.5 kg (23.2 lbs)	

Model	DXS18	DXS15mkII	DXS12mkII
System Type	Powered subwoofer, Band-pass type		
Components	LF	15" cone, 2.5" voice coil	12" cone, 2.5" voice coil
Frequency Range (-10dB)	32Hz - 120Hz	40 Hz - 150 Hz	42 Hz - 150 Hz
Output Power*1 (Dynamic)	1020W		
Maximum SPL (1m, on axis)	136dB SPL	135dB SPL	134dB SPL
Dimensions (W x H x D)	563 x 683 x 721 mm (22.2" x 26.8" x 28.4")	480 x 611 x 614 mm (18.90" x 24.06" x 24.17")	400 x 567 x 570 mm (15.75" x 22.32" x 22.44")
Net Weight	49.7kg (110lbs)	36.0 kg (79.4lbs)	30.0 kg (66.1lbs)

*1 Power rating (120V, 25°C). This is total value of individual output power.

Loudspeakers

Versatile, Rugged, and Ready for the Road

The CZR and CBR Series passive loudspeakers offer high power handling and outstanding sonic performance in portable, lightweight, yet rugged designs. For even more bass authority, CXS XLF Series passive subwoofers can extend your system's low frequency performance down to a startling 29Hz for solid low frequency reproduction that's sure to impress.



**CZR series,
CBR series**
Superior Power-Handling.
Professional Performance.

Key Features

Best in Class SPL performance



The CZR15 and CZR12 can both produce 129dB SPL with a power rating of 800W for the CZR15 and CZR12, and 700W for the CZR10 in passive mode. Every CZR model is also equipped with Biamp mode to maximize performance.

Custom Transducers



We thoroughly analyzed a wide range of transducers, choosing low-distortion units that provide extremely flat frequency response, wide dynamic range and high power handling.



Intelligent Processing Maximizes System Performance



When used in combination with Yamaha PX Series amplifiers or other compatible signal processors, CZR Series and CBR series loudspeakers can take full advantage of dedicated speaker processor tuning settings.

Premium Materials and Road-tested Toughness



The specially designed CZR series enclosure features lightweight, robust premium plywood with a rugged polyurea coating for improved portability and durability on the road.

Rugged, Portable Enclosure Design



CBR Series speakers feature a compact, durable, lightweight plastic cabinet equipped with newly designed ergonomic handles that greatly facilitate system transport and setup.

Versatility for a Variety of Applications



CZR and CBR Series speakers feature an optimized 50° wedge angle for floor monitoring as well as rigging points for rigged applications using standard eyebolts.

CXS XLF series

Extended Low Frequencies, Outstanding Clarity



Stunning Bass Performance



The CXS XLF Series offers two passive subwoofers: the CXS18XLF (18-inch) and the CXS15XLF (15-inch), both of which can extend system low-frequency response for truly impressive LF reproduction.

Durable, Portable, Road Ready



Enclosure exteriors feature a durable, damage-resistant polyurea coating to protect the cabinet. Optional casters are available for faster, more convenient positioning and setup.

Cardioid Mode



CXS-XLF subwoofers provide a cardioid mode that can be used in combination with Yamaha PX Series amplifiers or other compatible signal processors.

Specifications

Model	CZR15	CZR12	CZR10	
Speaker type	2-way Speaker, Bass-reflex			
Components	LF	15" Cone, 3" Voice Coil, Neodymium magnet	12" Cone, 3" Voice Coil, Neodymium magnet	10" Cone, 3" Voice Coil, Neodymium magnet
	HF	2" Voice Coil, 1" Throat Compression Driver, Titanium diaphragm, Neodymium magnet		
Frequency Range (-10dB)	Passive: 43 Hz - 20 kHz, Normal (*1): 34 Hz - 20 kHz, FOH (*1): 33 Hz - 20 kHz	Passive: 43 Hz - 20 kHz, Normal (*1): 42 Hz - 20 kHz, FOH (*1): 38 Hz - 20 kHz	Passive: 43 Hz - 20 kHz, Normal (*1): 46 Hz - 20 kHz, FOH (*1): 43 Hz - 20 kHz	
Power rating (EIA 426-A, Passive)	NOISE	400 W		350 W
	PGM	800 W		700 W
	MAX	1600 W		1400 W
Maximum SPL (1m, on axis)	129dB SPL		127dB SPL	
Floor Monitor Angle	50° Symmetrical		50°	
Enclosure Material, Finish, Color	Plywood, Durable Polyurea coating, Black			
Dimensions (W x H x D)	450 x 761 x 460mm (17.7" x 30.0" x 18.1")	410 x 646 x 394mm (16.1" x 25.4" x 15.5")	315 x 537 x 345mm (12.4" x 21.1" x 13.6")	
Net Weight	21.3kg (47.0lbs)	18.0kg (39.7lbs)	14.0kg (30.9lbs)	

*1: with recommended DSP processing.

Model	CBR15	CBR12	CBR10	
Speaker type	2-way Speaker, Bass-reflex Type			
Components	LF	15" Cone, 2.5" Voice Coil, Ferrite Magnet	12" Cone, 2" Voice Coil, Ferrite Magnet	10" Cone, 2" Voice Coil, Ferrite Magnet
	HF	1.4" Voice Coil, Compression Driver, Ferrite Magnet		1" Voice Coil, Compression Driver (Ferrite Magnet)
Frequency Range (-10dB)	46 Hz - 20 kHz	48 Hz - 20 kHz	50 Hz - 20 kHz	
Power rating (EIA noise)	NOISE	250 W	175 W	
	PGM	500 W	350 W	
	MAX	1000 W	700 W	
Maximum SPL (1m, on axis)	126 dB SPL	125 dB SPL	123 dB SPL	
Floor Monitor Angle	50° Symmetrical		50°	
Enclosure Material, Color	Plastic, Black			
Dimensions (W x H x D)	455 x 700 x 378 mm (17.9" x 27.6" x 14.9")	376 x 601 x 348 mm (14.8" x 23.7" x 13.7")	308 x 493 x 289 mm (12.1" x 19.4" x 11.4")	
Net Weight	17.7 kg (39.0 lbs)	13.9 kg (30.6 lbs)	9.4 kg (20.7 lbs)	

Model	CXS18XLF	CXS15XLF
Speaker type	Subwoofer, Bass-reflex	
Components	LF	15" Cone, 4" Voice Coil, Ferrite magnet
Frequency Range (-10dB)	Passive: 32 Hz - 3 kHz, Normal (*1): 32 Hz - 150 Hz, Extend (*1): 29 Hz - 150 Hz	Passive: 33 Hz - 3.5 kHz, Normal (*1): 35 Hz - 150 Hz, Extend (*1): 33 Hz - 150 Hz
Power rating (MAX, EIA 426-A)	2000 W	1600 W
Maximum SPL (Calculated)	133 dB *2	130 dB *2
Enclosure Material, Finish, Color	Plywood, Durable Polyurea coating, Black	
Dimensions (W x H x D)	550 x 657 x 720mm (21.6" x 25.9" x 28.4")	450 x 587 x 600mm (17.7" x 23.1" x 23.6")
Net Weight	43.6kg (96.1lbs)	35.6kg (78.5lbs)

*1: with recommended DSP processing. *2: Half space (2π)

Power Amplifiers

Intelligent Processing. Serious Power.

Since the 1976 release of the P2200, our first professional power amplifier, Yamaha has strived to achieve the highest possible levels of sound quality, power output, and reliability in sound reinforcement, studio, and installed applications. The new PX series power amplifiers continue this tradition of excellence, utilizing Yamaha's renowned digital signal processing technology and unparalleled professional audio expertise to achieve a light, durable design, capable of delivering maximum output power to the speakers while simultaneously protecting them with optimized processing. Driven by a newly-developed Class-D amplifier the four new models that make up the PX series are suitable for a broad range of sound reinforcement and installation environments.



Key Features

Powerful, Efficient Design



PX series models are driven by a newly developed Class-D amplifier engine that concentrates all the necessary functions into a single custom LSI chip, and uses PLL technology to control transfer characteristics in real time, achieving impressive sound quality and power. This new single-chip architecture offers higher output and improved performance in a more lightweight, yet reliable design. A combination of cutting-edge technology and an exhaustive component selection process make PX Series amplifiers supremely efficient amplifiers with low noise, reliable protection features, and above all, outstanding sound quality.

Easy, Intuitive Setup



To accommodate users with varying levels of sound reinforcement experience, the PX series offers both Basic and Advanced setup modes. Basic mode allows even inexperienced users to easily configure a system that meets their needs, while Advanced mode offers experienced users more precise control over every aspect of their sound. All models feature eight programmable presets for instant recall and reduced setup time that can be saved to a USB thumb drive and shared with other PX amplifiers.

Sophisticated Signal Processing



PX series models come with flexible onboard crossover, filters, delay, limiter and PEQ functions as standard, allowing quick and easy system setup via the LCD display, without the need for any additional outboard gear. All models also feature Yamaha's intelligent D-CONTOUR multi-band dynamic processing, allowing users to achieve consistent clarity and high-quality sound for either front-of-house or monitor configurations.

Flexible Connectivity, Durable Construction



Equipped with both XLR and TRS inputs, and binding post, SpeakON and phone outputs, PX Series amplifiers offer the flexibility needed for a variety of setups and environments, and the durability to stand up to extended road-use thanks to the sturdy chassis and carrying handle design.

Smart Configuration Wizard



Speaker Type

The Config Wizard allows users to save even more time during system setup by assigning optimized speaker settings that match your system configuration— simply select the appropriate filter and crossover for each speaker, and Config Wizard will do the rest. Config Wizard also features presets* specifically tuned for each model in Yamaha’s extensive lineup of passive speakers, including the popular CBR, Club, Installation, and VXS/VXC series, and an advanced mode that allows more detailed configuration of parameters such as input routing, input sensitivity, amp gain, and speaker impedance.

*Presets may be added and updated in the future.

Versatile Lineup

With a lineup comprising four new models – 1,200 W x 2 PX10, 1050 W x 2 PX8, 800 W x 2 PX5, and 500 W x 2 PX3* – the PX series offers a versatile range of different power output capabilities suitable for a wide variety of sound reinforcement applications. The PX3 and PX5 also come equipped with a Power Boost Mode that combines the power output from two channels into one, achieving significantly higher single-channel power levels to drive high power handling speakers when needed (note that the number of channels available is halved in this case).





*Power output figures rated at 4 Ω

Utilities for Safe and Secure Operation



PX Series amplifiers make it fast and easy to configure panel settings, monitor amplifier status, and transfer data via USB, and also feature a panel locking function to keep your settings safe. Users can even export information such as internal device and log data to USB for rapid troubleshooting should difficulties arise.

Specifications

Model					
		PX10	PX8	PX5	PX3
Output power: 1kHz, Non-clip, 20msec Burst, Both channels driver	8 Ω	1000W x 2	800W x 2	500W x 2	300W x 2
	4 Ω	1200W x 2	1050W x 2	800W x 2	500W x 2
	2 Ω	700W x 2	600W x 2	500W x 2	300W x 2
	8 Ω / Power Boost Mode	—	—	800W x 1	600W x 1
	4 Ω / Power Boost Mode	—	—	1400W x 1	1000W x 1
Amplifier type (Output circuitry)		Class D, Balanced output circuit (BTL)			
THD+N	1kHz, 10W	0.1%			
	1kHz, Half power	0.3%			
Frequency Response		1W, 8Ω, 20Hz to 20kHz			
Crosstalk		Half Power, 8Ω, 1kHz, Vol max input 150Ω shunt			
Dimensions (W x H x D)		480 x 88 x 388mm (18-7/8" x 3-7/16" x 15-2/8")			
Net Weight		7.4kg (16.31lb)	7.2kg (15.87lb)	6.9kg (15.21lb)	6.9kg (15.21lb)

Portable Sound Systems

All the World's YOUR Stage

When it comes to portable live sound solutions, it doesn't get any better than STAGEPAS. With impressive power and sound quality, intuitive control, and a comprehensive array of flexible features, including a new Bluetooth® wireless audio streaming function, this all-in-one PA system builds on the STAGEPAS legacy of providing everything you need to transform any environment into your own personal stage.



STAGEPAS 400BT/ 600BT

All the World's YOUR Stage

Fast and Easy Setup



With its intuitive, simple design, STAGEPAS can be set up and ready to go in a matter of minutes. Even users with no PA operation experience will be able to produce high-quality sound in a few minutes.

PA for Any Occasion



Flexible enough to be used in a truly impressive range of environments and applications, STAGEPAS systems can be set up in a variety of PA and monitoring configurations.

1-Knob Master EQ™



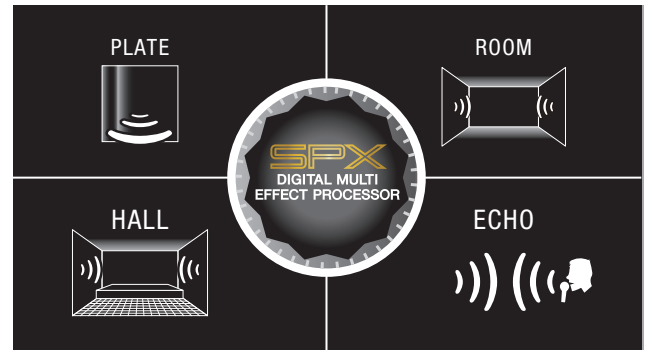
This new function gives you instant mastery of your mix, allowing you to match your sound to your surroundings. Simply turn the knob to optimize the EQ for speaking engagements, musical performances, or just to kick in more bass; getting your sound right has never been so easy.

Bluetooth



Equipped with a Bluetooth input for wireless audio streaming from smartphones or tablets, the Stagepas BT models let you seamlessly play backing tracks for gigs, or set the tone of parties, conferences, or events with the audio playback of your choice.

SPX Digital Reverb



One knob gives you access to four different high-quality reverb setting, perfect for vocals and acoustic instruments. Simply turn the knob to change the reverb type and time, and then you're ready to apply just the right amount of high-resolution effect to each channel.

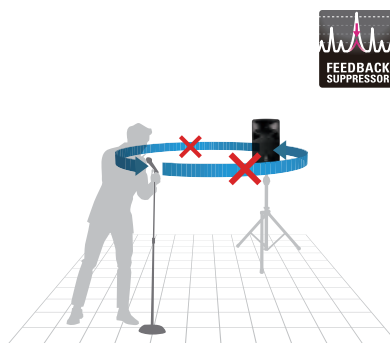


Ultimate Portability



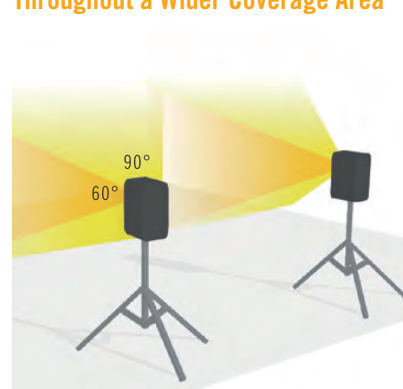
Compact and light enough to carry by hand, STAGEPAS is an all-in-one PA system that you can take literally anywhere. The two lightweight speakers and a detachable mixer, along with one pair of speaker cables and a power cord, combine into an attractive, compact package that provides the kind of portability that no roadie would ever dare to dream of.

Feedback Suppressor





Avoid the uncomfortable feedback that plagues so many sound reinforcement novices and professionals alike. Usually found only on high-end gear, this intelligent function removes feedback automatically with the push of a button, giving you a clear, professional sound.

A Consistent Sonic Experience Throughout a Wider Coverage Area



For use in larger spaces with a sizable audience, STAGEPAS is capable of delivering consistent high-quality sound throughout a greater coverage area than ever before with custom-designed speaker components that ensure the entire audience shares the same sonic experience.

Specifications

Model	 STAGEPAS 600BT	 STAGEPAS 400BT
System Type	Powered mixer with 2 passive speakers	Powered mixer with 2 passive speakers
Mixing channels	10	8
Analog inputs	4 mono mic / line + 6 mono / 3 stereo line	4 mono mic / line + 4 mono / 2 stereo line
Bluetooth Connection	Bluetooth Ver.4.1, A2DP v1.2, Maximum Range: Approx. 10 m	Bluetooth Ver.4.1, A2DP v1.2, Maximum Range: Approx. 10 m
Channel EQ	3-band (HIGH Shelving: 8kHz, MID peaking: 2.5kHz, LOW Shelving: 100Hz)	2-band (HIGH Shelving: 8kHz, LOW Shelving: 100Hz)
Power Rating (Dynamic)	680W (340 W + 340 W)	400W (200W + 200W)
Speaker Transducers	LF: 10" (25cm) cone, HF: 1.4" (3.56cm) voice coil compression driver	LF: 8" (20cm) cone, HF: 1" (2.54cm) voice coil compression driver
Dimensions (W x H x D)	Speaker: 335 x 545 x 319mm, Mixer: 348 x 197 x 135mm	Speaker: 289 x 472 x 275mm, Mixer: 308 x 180 x 116mm
Net Weight	25.6 kg (56.4lbs) (2 x Speaker 10.9 kg + Mixer 3.8 kg)	18.3 kg (40.3 lbs) (2 x Speaker 7.7kg + Mixer 2.9kg)

Studio Monitor Speakers

Accuracy is Everything

Yamaha studio monitors are designed for professional engineers who are serious about their mixes and demand the most precise, high-quality sound for their monitoring needs. Accuracy is our top priority throughout the entire production process, from the meticulous material selection for our own customized transducers, to the rigorous testing of the acoustic properties and design of the enclosure. Combined with built-in high-efficiency amplifiers our near-field monitors deliver a supremely accurate and honest sonic platform to build your sound upon, in a powerful yet compact package that meets the demands of even the most discriminating ear.

HS Series

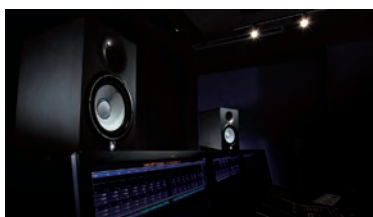
TIME FOR A REALITY CHECK



Ever since the 1970's the iconic white woofer and signature sound of Yamaha's nearfield reference monitors have become a genuine industry standard for a reason - their accuracy. Unlike studio monitors with added bass or treble frequencies which may sound more flattering at first, HS Series speakers were designed to give you the most honest, precise reference possible, providing an ideal sonic platform to build on throughout the mixing process.

HS Studio Monitor Sound Philosophy

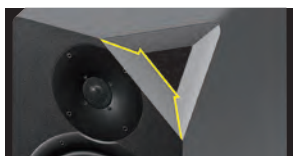
When selecting studio reference monitors for mixing and music production, one consideration takes precedence above all others - accuracy.



Beginning with the legendary NS-10M, and continuing with the MSP and previous HS Series, this new lineup shares a design philosophy that emphasizes sonic purity without any coloring or alternation of the original sound. While many manufacturers seek to enhance the sound of their monitors to make them sound impressive, Yamaha studio monitors are designed to achieve the best possible mix by ensuring that the sound from your speakers is true to the original. HS Series nearfield reference monitors are the embodiment of this philosophy, delivering a concise sound image, high resolution, and flat response.

Low Resonance Enclosure Design

HS Series cabinets are designed to eliminate unwanted resonance and maximize the sound reproduction accuracy. The enclosures are constructed from a dense and resilient MDF with a damped acoustic response that is perfectly suited for reference monitors.



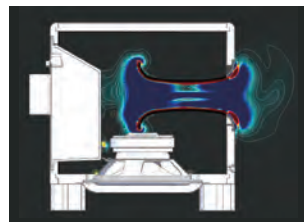
Drawing from over 100 years of Yamaha piano design experience, HS Series engineers also employed a traditional three-way mitered-joint technique. This construction firmly anchors the corners of the enclosure to dramatically improve durability and eliminate unwanted resonance that can influence overall sound.

High Performance Amp Unit

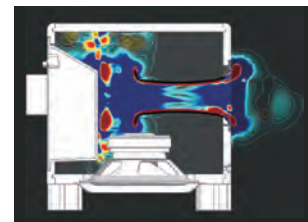
HS Series speakers employ an amplifier unit perfectly matched to the transducers utilized in each model of the series. Featuring a bi-amp design with a separate dedicated amp for both the woofer and the tweeter, this amplifier unit ensures that each HS Series speaker consistently delivers high-resolution sound with exceptionally flat response across the sound spectrum.



Cutting Edge Noise Reduction Technology



▲ HS85



▲ Example :HS10W speaker port

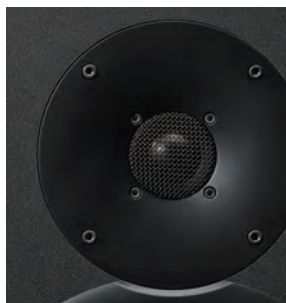
Speaker port design can greatly influence overall sound clarity. Often a vortex occurring at either end of the port will generate air vibrations inside of the port, causing unwanted noise. Advanced Yamaha noise reduction technology effectively controls airflow and suppresses noise-generating vortices.

Efficient Transducers

The HS Series features newly developed transducers that achieve astonishingly smooth response over a wide bandwidth. These transducers utilize an advanced magnetic field design that regulates the magnetic field for provide seamless, natural sonic transitions. Materials and design were carefully re-examined and optimized for drastically improved signal reproduction accuracy of throughout the audio spectrum.

● Tweeter

The HS Series features a new high-efficiency 1" dome tweeter that extends the usable frequency range considerably and delivers superb sonic resolution. Utilizing a wave guide designed to minimize vibration, the new tweeter delivers smooth, distortion-free high end up to 30 kHz.



● Woofer

With newly designed, ultra-responsive woofers that utilize large, carefully selected magnets, HS Series monitors achieve low distortion with a well-defined low end at any output level. All of the components in these high-power woofers, including the woofer ring and basket, further contribute to stunning bass while providing clear, accurate midrange reproduction.



Controls and Connectivity

HS Series full-range models feature two response controls with detailed settings that allow them to adapt to the acoustics of various rooms. ROOM CONTROL attenuates the unnatural, exaggerated low-end that can occur when speakers are placed next to walls, while HIGH TRIM allows for more flexible high frequency response control. HS full-range speakers can accommodate a wide range of balanced and unbalanced sources such as mixers, keyboards, and audio interfaces with XLR and TRS phone jacks. With the HS8S subwoofer, a HIGH CUT control sets the subwoofer output cutoff frequency from 80Hz to 120Hz, while a PHASE switch adjusts the phase of the subwoofer output. A LOW CUT switch and LOW CUT control enable attenuation of low frequencies between 80Hz and 120Hz. The HS8S subwoofer is equipped with XLR connector and TRS phone jack inputs, and XLR connector outputs.



Flexible Setup (HS-I only)

Mounting points make the HS-I models compatible with optional wall, ceiling, and baton brackets, adding the mounting flexibility needed for a broad range of sonic environments.



Specifications

Model	HS5 / HS5W HS51 / HS51W		HS7 / HS7W HS71 / HS71W		HS8 / HS8W HS81 / HS81W		HS8S
System Type	2-way bi-amp powered speaker, Bass-reflex type						Powered subwoofer, Bass-reflex type
Components	LF	5" cone				8" cone	
	HF	1" dome					
Frequency Range (-10dB)	54Hz - 30kHz		43Hz - 30kHz		38Hz - 30kHz		22Hz - 160kHz
Output Power*1 (Dynamic)	70W		95W		120W		150W
Connectors	INPUT: XLR3-31 type (balanced), PHONE (balanced)						INPUT: XLR3-31 type (balanced), PHONE (balanced) / OUTPUT: XLR3-32 (balanced) x 2 (L&R)
Cabinet Material, Color	MDF, Black / White						
Rigging Points	HS51: Four surfaces x 2 x M5 (60mm pitch)		HS71: Four surfaces x 2 x M8 (120mm pitch)		HS81: Four surfaces x 2 x M8 (120mm pitch)		—
Dimensions (W x H x D)	170 x 285 x 222mm (6.7" x 11.2" x 8.7")		210 x 332 x 284mm (8.3" x 13.1" x 11.2")		250 x 390 x 334mm (9.8" x 15.4" x 13.1")		300 x 350 x 389mm (11.8" x 13.8" x 15.3")
Net Weight	HS5: 5.3kg (11.7 lbs.), HS51: 5.5 kg (12.1 lbs.)		HS7: 8.2kg (18.1 lbs.), HS71: 8.7 kg (19.2 lbs.)		HS8: 10.2kg (22.5 lbs.), HS81: 10.7 kg (23.6 lbs.)		12.5kg (27.6lb)

Mixer with USB Audio Interface

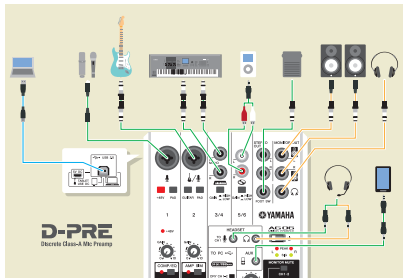
Create, Record, and Share

Ready to share your musical creations with the world? AG series mixing consoles are ideal tools for fast, easy production of musical content destined for live streaming sites like YouTube LIVE, online lessons in real time, or video or audio podcasts created offline. Studio-quality sound will give your content a professional edge that can help to build and expand your fan base.



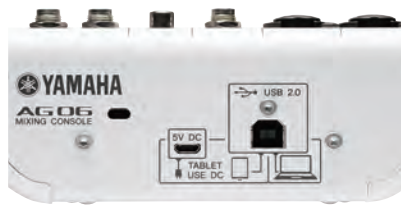
Key Features

Flexible Connectivity



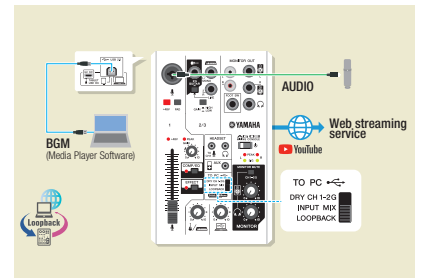
Switchable phantom power and a switchable Hi-Z (GUITAR) input allow you to connect a variety of microphones, instruments and external audio players directly to your console.

High-resolution (192kHz / 24bit) Playback and Recording



High-resolution reproduction delivers every nuance of your performances and recordings with flawless fidelity and accurate imaging for music production, mixing, broadcasting, and high-definition listening.

LOOPBACK for Live Streaming



A LOOPBACK function enables live broadcasting of all the microphones, instruments, and other audio sources along with background music from your computer or tablet.

Bundled Software

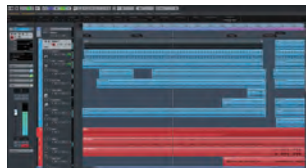
AG DSP Controller

Compatible with Windows and Mac computers, AG DSP Controller gives you additional control of 1-TOUCH DSP parameters such as COMP, EQ, HPF, reverb and the AMP SIM on your AG mixing console.



CUBASE AI

The streamlined version of the most popular digital audio workstation for creating, recording and mixing music.





CUBASIS LE

The streamlined version of the ultimate music workstation for the iPad.



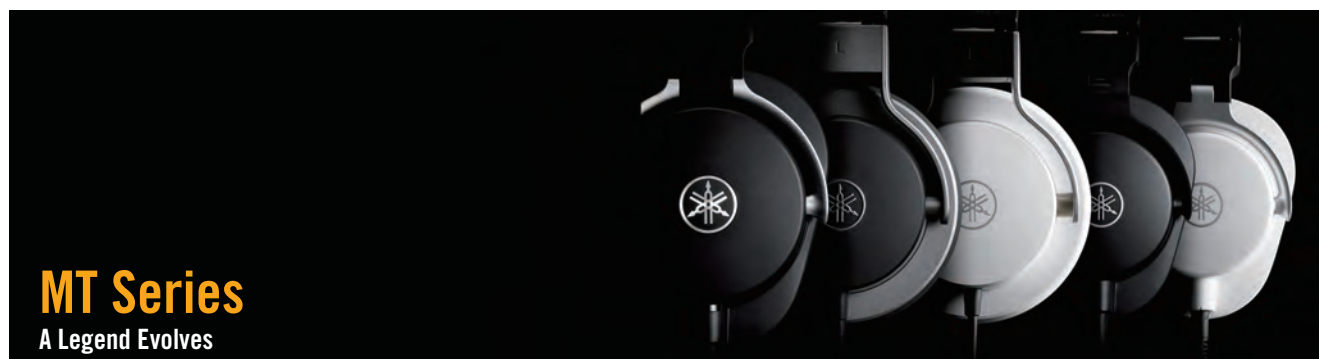
Specifications

Model		 AG06	 AG03
Inputs	Mono[MIC/LINE]	2 including HEADSET MIC (Plug-in Power)	1 including HEADSET MIC (Plug-in Power)
	Stereo[LINe]	2	1 including Guitar input (Mono)
	AUX	1	1
Outputs	STEREO OUT	1	—
	MONITOR OUT	1	2
	PHONES	2 including HEADSET PHONES	2 including HEADSET PHONES
	AUX SEND	—	—
BUS		Stereo: 1	Stereo: 1
DSP		CH1: COMP / EQ, EFFECT (SPX Reverb) CH2: AMP SIM, EFFECT (SPX Reverb)	CH1: COMP / EQ, EFFECT (SPX Reverb)
USB		USB Audio: 2 IN / 2 OUT, USB Audio Class 2.0 compliant, Sampling Frequency: Max 192 kHz, Bit Depth: 24bit	
Power requirements		USB bus powered	USB bus powered
Bundle software		Cubase AI/AG DSP Controller	
Dimensions (W x H x D)		155 x 63 x 202mm	129 x 63 x 202mm
Net Weight		1.0kg	0.8kg

Headphones

A Legend Evolves

From the NS-10M to the HS Series, Yamaha has remained faithful to a design philosophy focused on delivering sonic accuracy that allows professionals to develop and establish their own signature sound. MT Series studio monitor headphones remain true to this fundamental concept, delivering precise sound reproduction that meets the demanding requirements of today's professional studio and monitoring applications.



MT Series

A Legend Evolves

Key Features

Tools that Professionals Trust



The MT series inherits the design philosophy behind Yamaha studio monitor speakers, delivering faithful reproduction of the source signal.

Superb Isolation. Premium Comfort.



HPH-MT series headphones provide superb comfort and isolation, and are specifically designed to meet the high standards of professional users.

High Resolution Sound



MT series headphones feature custom drivers equipped with CCAW voice coils (MT8 = 45mm, MT7 and MT5 = 40mm) and powerful neodymium magnets to achieve precise definition and control.

Specifications

Model	 HPH-MT8	 HPH-MT7 / HPH-MT7W	 HPH-MT5 / HPH-MT5W
Type	Closed-back, Circumaural (Over Ear)		
Frequency Response	15Hz - 28kHz	15Hz - 25kHz	20Hz - 20kHz
Sensitivity (SPL)	102dB/mW	99 dB/mW	100dB/mW
Maximum Input Power	1600 mW (at 1kHz)		
Impedance	37 Ω (at 1kHz)	49 Ω (at 1kHz)	51 Ω (at 1kHz)
Driver Unit	45 mm, Dynamic, CCAW Voice Coil / 40 mm, Dynamic, CCAW Voice Coil		
Cable	1.2 m (3.9ft), coiled cable; 3.0 m (9.8ft), straight cable / 3.0m (9.8ft), straight cable		
Connector	3.5 mm (1/8") stereo with 6.3 mm (1/4") stereo adaptor		
Net Weight	350g (0.8lb) (Without cable&plug)	360g (0.8lb) (With cable&plug)	360g (0.8lb) (With cable&plug)



SHARING PASSION & PERFORMANCE

YAMAHA CORPORATION

P.O.BOX1, Hamamatsu Japan

www.yamaha.com/proaudio

*All specifications are subject to change without notice. *All trademarks and registered trademarks are property of their respective owners.