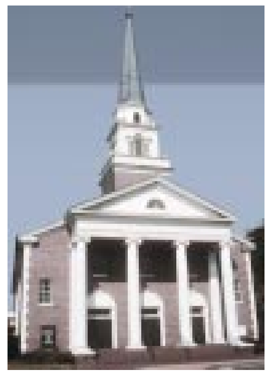
Automatic Mixer Application Guide



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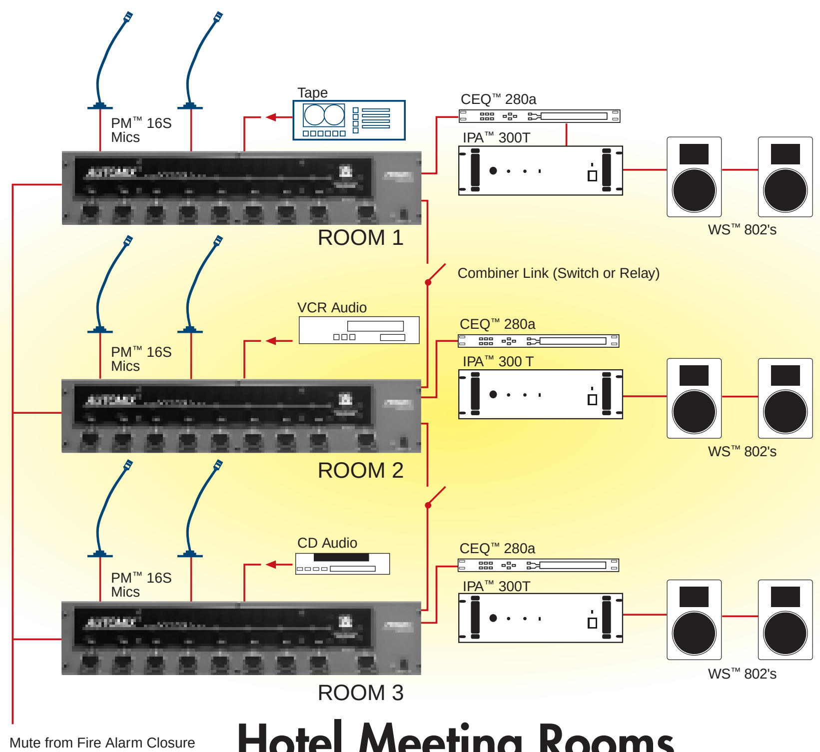
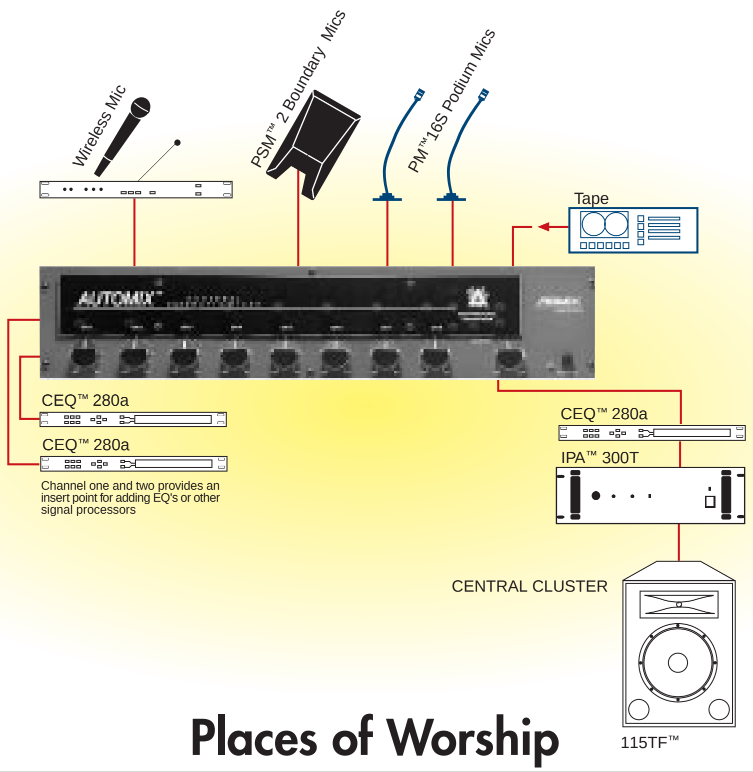


Hotel Meeting Rooms



Places of Worship

Automatic Mic Mixer Fundamentals

Automatic mic mixers are very valuable to any sound system, whether you are designing or installing a sound system into a church, courtroom, boardroom, conference center, council chamber, classroom, auditorium, or broadcast facility. Automatic mixers benefit all of these sound system applications by reducing feedback, increasing intelligibility, and producing cleaner, louder sound. Most auto mixers have very different methods of operation.

When working with auto mixers, the system designer should have a strong working knowledge of acoustics and electronics. In this first section, we discuss the terminology required to build a strong base for designing sound systems with auto mixers. We also discuss the operation of various auto mixers with real-world design applications.

Level vs. Gain

LEVEL - refers to the sound-pressure level or the audio signal voltage in the system at a given moment, typically measured with a SPL meter or a VU meter, respectively.

A clear understanding of the difference between gain and level is extremely important. Audio engineers have a habit of using these terms interchangeably. But to fully understand how auto mixers operate, we have to make sure the definitions are applied correctly.

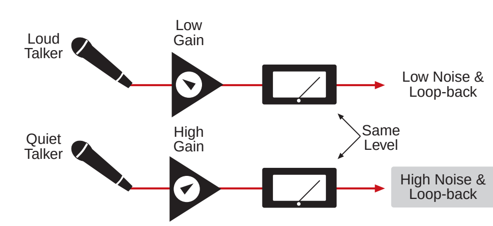
GAIN - amount of amplification in a signal path between the input and the output.

As mentioned earlier, knowledge of acoustics is required to understand the difference between electrical gain and acoustical gain. Both are very important to the design of a trouble-free system.

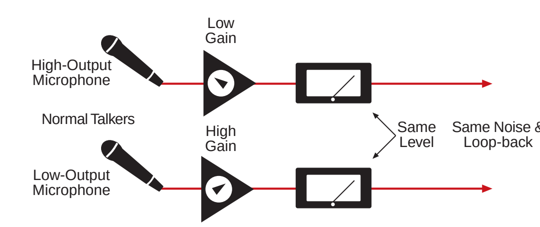
Electrical gain - is the amount of amplification in the channel between the input and the output. Electrical gain changes when the mixer controls are changed. This gain is indicated by the settings of the controls on the mixer.

Acoustical gain - is the relationship between natural and amplified sound. It is measured by using a SPL meter and observing the difference between the level with the microphone on and the level with the microphone off.

Some of the confusion about using gain and level interchangeably is the illusion that one can't be changed without the other. Every audio system has three parts: an input level, system gain, and an output level. Let's look at what this means when dealing with microphones, mic preamps (the gain device), and some VU meters (the level device) to illustrate the difference between electrical gain, acoustical gain, and level.



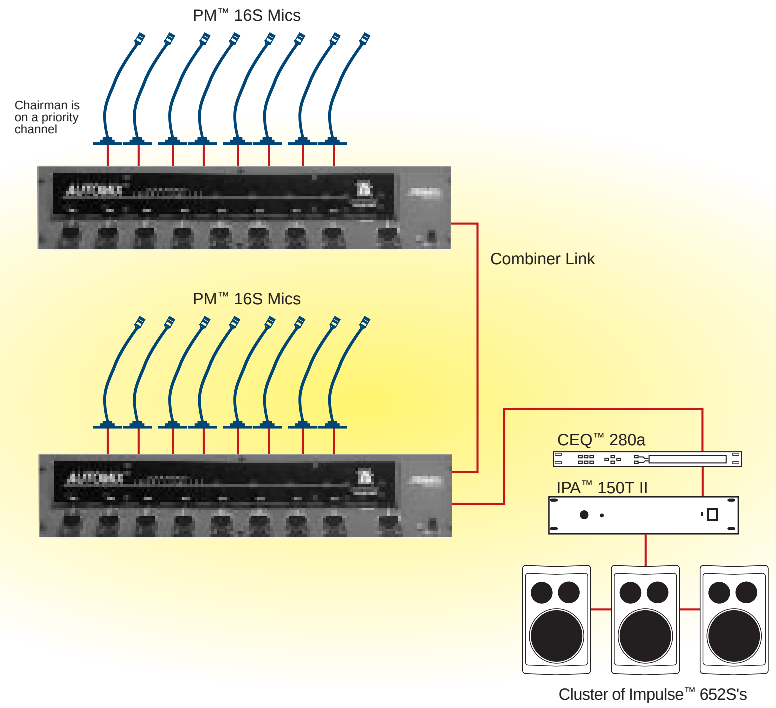
In Fig. 1 above, a loud talker is using channel one of our simple system and a quiet talker is using channel two. To make the output level on the VU meters match, we will have to add more electrical gain to the quiet talker. Doing this makes the output level of channel one and two equal. However, the electrical gain and the acoustical gain are not equal between the channels. The result is a higher noise output of channel two and a greater danger of feedback. Why? Because we increased the electrical gain of the mic preamp and not the acoustical level of the quiet talker, which amplified the background noise and increased the ambient loop-back into the microphone.



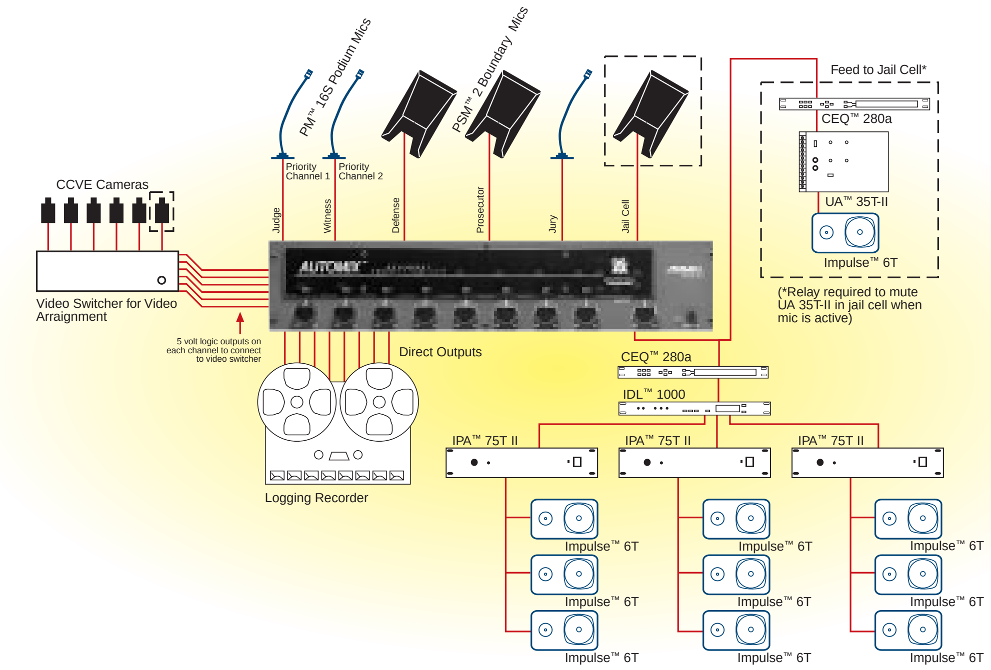
In fig. 2 above, people are talking normally (one does not appear to be any louder or softer than the other), but we have one microphone with a high-output and another with a low-output. The electrical gain of the low-output microphone channel will have to be raised so that both channels have the same output level. The result is the same level and acoustic gain but different electrical gain, which produces an "even" output. It is vital to keep these definitions clear when working with auto mixers: level is the amplitude (or size) of the signal and gain is the amount of amplification.

Feedback

One of the key reasons auto mixers have become popular in sound systems is because they assist in preventing acoustic feedback. Acoustic feedback refers to the introduction of acoustic energy from a sound system back into the same system. Auto mixers employ a function called NOM to reduce acoustic feedback caused by the additional gain of active mic channels that are not being used.



Council Chambers



Court Rooms

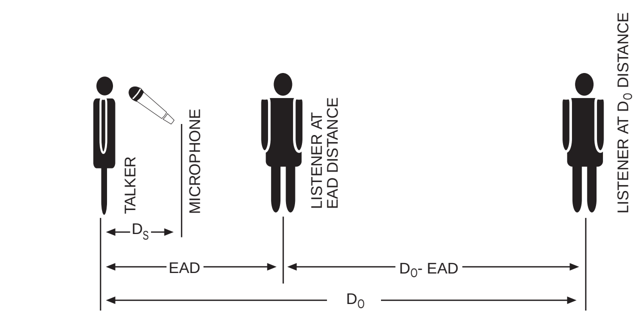
<u>NOM</u> - Number of Open Microphones provides a method of keeping a sound system from going into feedback when more than one microphone is active. Since each microphone is sampling the sound field, every time you double the active microphones (or NOM), you will have increased the system gain by 3 dB. This additional gain can instantly put the system into feedback. NOM compensation provides a means to avoid this by simply reducing the overall system gain by 3 dB every time the number of open microphones is doubled.

NOM in dB = 10 log (NOM)

<u>NAG/PAG</u> - Needed Acoustical Gain and Potential Acoustic Gain. To effectively design a sound system, you will need to know the mathematical formulas for both NAG and PAG. NAG/PAG will tell you in advance if the system you are designing will have enough Potential Acoustic Gain before feedback. As shown by the formulas, NOM plays a very important part in a properly designed sound system.

$$\text{NAG} = 20\log\left(\frac{D_0}{D_s}\right) - 20\log\left(\frac{EAD}{D_s}\right)$$

$$\text{PAG in dB} = \Delta D_0 - \Delta EAD = 10\log(\text{NOM}) + 6\text{ dB FSM}$$



Ds= Distance from the talker to the microphone

D_0= Distance from the talker to the farthest listener

EAD= Equivalent Acoustic Distance

FSM= Feedback Stability Margin

Automatic Mic Mixer Technology

How well an auto mixer performs is determined by **a)** *the decision method;* **b)** *its "switching" method; and* **c)** *how it performs NOM functions.*

Decision Making

All automatic mic mixers have to first determine which mic or mics need to be active. There are several different decision making methods on the market today. All of the different variations are based on either a fixed threshold or an adaptive threshold technology. Each of these decision making variations will be discussed as to their success in the field.

The first technology used in an auto mixer is the fixed threshold. The most widely used fixed threshold technology is the VOX (voice activated switch). A VOX uses technology found in inexpensive speakerphones. VOX systems activate the mic as soon as the ambient level reaches a predetermined threshold. You probably have noticed that when talking to someone who is using a speaker-phone, it is silent when no one is speaking. But as soon as someone drops a pen on the table, you hear "room" noise for a brief second, then it returns back to total silence. The impact of the pen hitting the table was above the threshold set in the VOX, turning the mic on, but the "room" noise was not loud enough to keep it on. This technology doesn't work in everyday audio applications. If a VOX-type auto mixer was used in a church, for instance, all of the microphones would exceed the threshold, (putting the system into feedback) when the congregation started to sing.

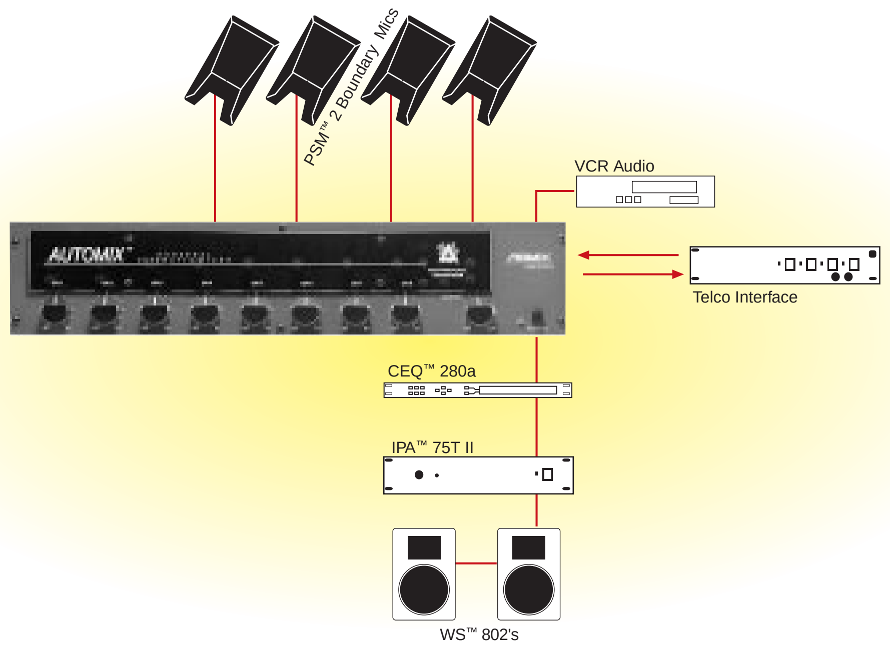
This scenario is why most systems today use a variation of adaptive threshold technology for auto mixer decision making methods. Adaptive threshold takes dynamic ambient level changes into consideration, unlike fixed threshold methods which do not allow for this. There are various methods of adaptive threshold including ambience sensing, direction sensing, hi-to-low scanning, low-to-hi scanning, and gain sharing.

Ambient sensing systems use one or more microphones for "measuring" the ambient level. When one microphone is used in this way, it is listening to the overall level of the room and using electronics to set the threshold based on the ambience of the room. If the signal level at the microphone is stronger than the signal "set" by the ambient sensing mic, it will become active. The acoustic energy at the microphone will be stronger, with someone speaking directly into it, than the energy of the ambient noise at the sensing mic.

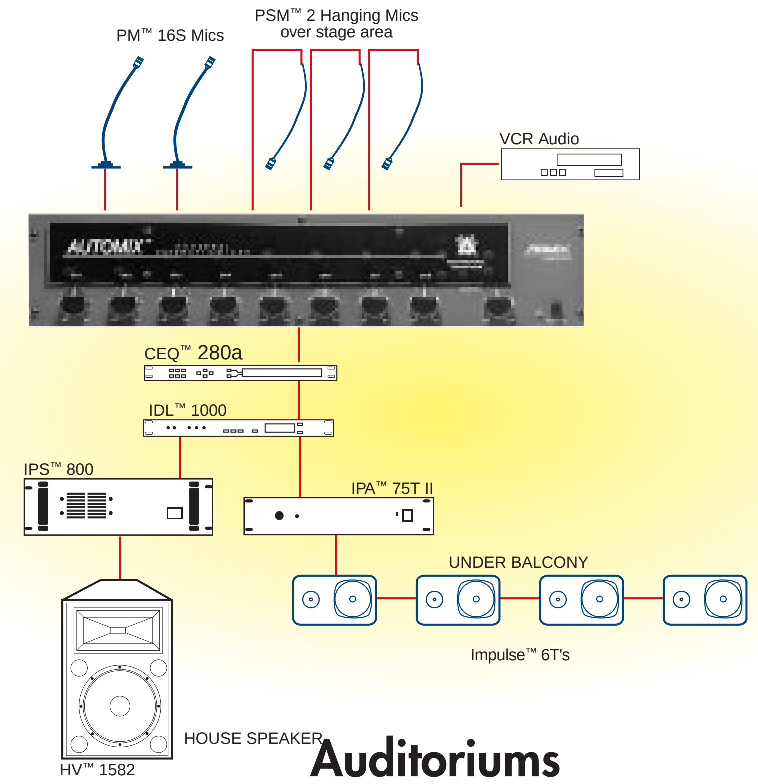
Another method of using ambient sensing is commonly referred to as Direction Sensing. Direction Sensing operates by comparing the talker's mic with the "ambient" mic. It basically has two microphones at each talker's location: one pointing to the talker and the other pointing in the opposite direction. The microphone pointing in the opposite direction of the talker is to "measure" the ambient noise level. The talker's mic becomes active when its level is greater than the level of the ambient mic.

Both methods of "sensing" work but become very costly to implement because of the need for "non-functioning" hardware. Adding extra microphones becomes a very expensive way to "automate" a mixer.

The most cost-effective and practical way of using adaptive threshold technology is gain sharing. Gain sharing uses the microphones that are required for normal operation and provides the "sensing" by



Boardrooms



Auditoriums

"splitting" the signal into two buses, an audio and control bus. The audio bus sends its signal to the output, which goes to a loudspeaker or recording device. The "audio signal" on the control bus is filtered and used to provide "intelligence data" for sharing the gain between inputs so that the system gain remains constant.

Switch Method

After the auto mixer determines which mic to activate, it must have a way to "switch" it on. This is known as the switch method.

Automatic mic mixers utilize a variation based on one of two "switching" schemes. The first and most widely used is a noise gate (not necessarily the best way, but usually the most cost effective.) The second method is by using a device called a downward expander.

A noise gate (or gate) permits signals above its threshold to pass through the device; however, if they fall below the threshold, the gate will not allow the signals to pass.

Gates are a commonly used method of switching microphones on. However, basic gate switching can be undesirable due to its tendency to "pop" when switched on, in addition to its employment of a fixed threshold decision making method. A gating technology was developed for auto mixers to solve "switching noise" problem. This is refered to as "gate at zero crossing." Mixers utilize this gate technology when the waveform crosses the zero amplitude axis (zero crossing), which dramatically reduces the noise artifacts of a basic gate. In addition to this "zero crossing" method, there are additional electronics and logic used with this technology to provide a more intelligent way to discriminate between ambient noise and a "real" signal.

A downward expander is an amplifier whose gain decreases as its input level is decreased. It works like this: say you were told to watch the input level of a preamp on a channel of a mixer, so that when the input VU level dropped to -3 dB on the input, you could reduce the preamp gain control to make the VU level on the output of the preamp -6 dB—and you were to continue this "downward" gain reduction as long as the signal level to the input of the preamp dropped. This is how a "downward expander" works when the threshold is -3 dB with a 1:2 ratio. The reason they are called expanders is because they "expand" the dynamic range by reducing the noise floor of the signal *while* the gain is decreasing.

Gain sharing provides a total system gain. The Peavey Automix uses a unique gain management system that combines gain sharing with downward expanders, utilizing the latest in VCA technology. When one person is talking into a microphone, the mic gets all of the gain available. When two people talk into two microphones, they share the available gain. This technique compares the needed gain with available gain to provide a seamless on and off transition of active microphones, while keeping the proper total system gain within the feedback stability margin.

NOM Method

Now that the mixer has decided what mic(s) should be active and how it will be activated, the next task is to determine how to keep itself from going into feedback. NOM requires an electronics method to determine how many mics are active and the method of how it will reduce the gain.

Two methods of providing NOM attenuation in auto mixers are linear and exponential.

Linear reduces system gain a fixed amount each time a microphone becomes active.

Exponential reduces system gain each time the number of active microphones double.

The three popular electronic methods for determining the number of active mics include VCA, digital master attenuators, and mix bus loading.

VCA technology is common in gain sharing systems because the VCA's side chain comes from the control bus described earlier. When using the VCA in this manner, it provides the two functions that make this type of technology extremely effective: expanding and a means by which NOM computations can be made.

Digital master attenuators use logic circuits (or microprocessors) to determine how many channels are active, then they reduce the gain either linearly or exponentially.

The third and last method is mix bus loading, which is a cost-effective way of providing NOM. This method puts a resistive load on the mix bus for every channel that has been "gated." The resistive load looks like a master volume control to the mix bus and "turns" down the volume linearly every time it "sees" an active channel.

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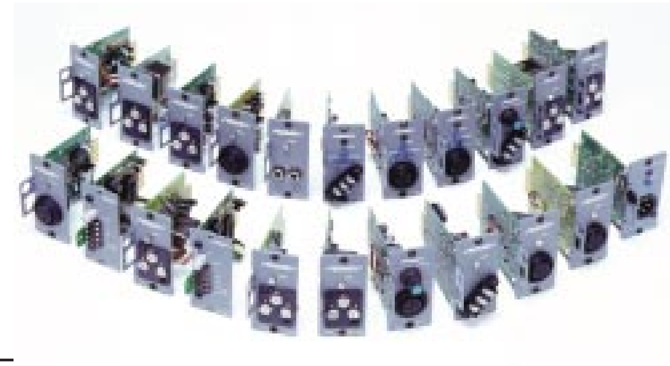
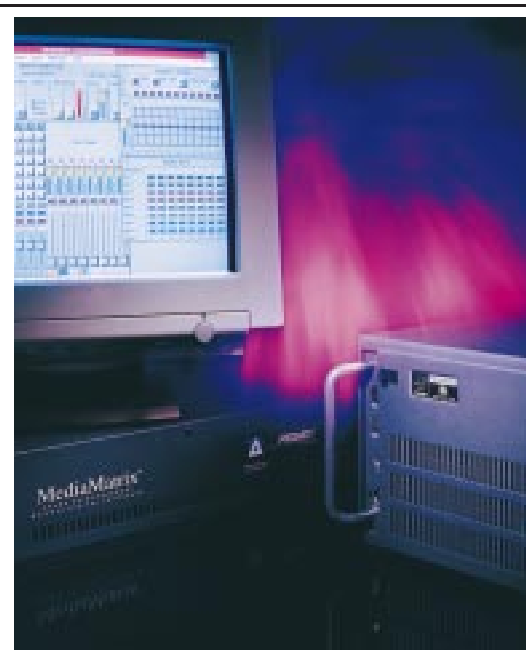
Peavey provides three auto mixer topologies for system designers.



1- AUTO modules for the MA and MMA Series mixer/amplifiers and preamp. This system uses a module that provides a gate with mix bus loading for cost-sensitive applications.

2-The Automix™ is a highly effective automatic system that uses Peavey's gain management system utilizing "downward" expanders and special calculations for NOM, making this a full-featured, stand-alone auto mixer at an extremely affordable price.



3- The MediaMatrix® MWare Version 1.2 (or greater) provides a complete auto mixer "toolbox" which has numerous design choices available. The MWare, running in a MediaMatrix Mainframe, MediaMatrix Miniframe, or the "Microframe" Digital Processor, gives the designer a complete digital sound system for any size job.

Peavey manufactures a complete line of microphones, signal processors, amplifiers, and loudspeakers for all applications.

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