Tutorial

Application of Automatic Mixing Techniques to Audio Consoles

By Dan Dugan

Automatic microphone mixers have become common in unattended sound reinforcement systems. Though many of these use simple on-off gating, continuous analog functions are also available that deliver better performance in demanding applications. A brief review of the field of automatic mixing functions is given. A prototype of an eight-channel controller has been built that provides automatic mixing functions as an automatic assistant to operators of multimike live sound mixing consoles. Feedback, ambient noise, and reverberation pickup are reduced, and mixer reaction time is eliminated in fast-moving discussions.

Why automatic mike mixing? Whenever multiple microphones are used, the quality of the audio is degraded in two ways. First, having more than one mike open at a time increases feedback from stage monitor and sound reinforcement speakers, and also increases the amplification of ambient noise. The increase in feedback and/or noise follows the curve shown in Fig. 1. Second, the pickup of a single sound source by multiple mikes causes a distortion of the tone quality called comb filtering, as shown in Fig. 2.[1]

Sound operators have to pull down the faders of idle mikes to avoid feedback, noise buildup, and comb filtering. In many situations it is hard to keep up with the action, and the operator has to make a compromise between two undesirable conditions. Either cues will be missed when an unexpected speaker or performer chimes in, or more mikes will have to be kept up than the operator would like for the best sound. It would appear that a gate, a voice-operated switch (VOX), might be a great help in this situation. Unfortunately, this simple technology is inadequate.

A VOX, or noise gate, has a fixed threshold, the sound level at which the channel opens up. The threshold may be manually adjustable, but it is fixed in terms of not reacting to the changing acoustic situation. In too many situations, if the threshold is set low enough to turn the gate on reliably, unwanted environmental sounds such as applause or music will exceed the threshold and turn the channel on as well. Figure 3 simulates the action of an ordinary noise gate.

Besides the threshold problem, there is another major disadvantage in multimike VOX systems. When applause turns all the mikes on, the system will go into feedback unless all channels are set at a very conservative (and probably inadequate) gain. The operator must keep one hand on the master fader.

The challenge for the development of automatic mike mixers was then twofold: first, to find a method for detecting when a mike is in use, while rejecting ambient noise; and second, to control the master gain so that feedback is avoided at the same time the maximum possible gain is available from each mike. A number of workable solutions to these problems have appeared since the debut of automatic mixers for sound reinforcement in 1975.[2]

The Elements of Automatic Mixer Action

It is helpful to analyze the workings of automatic mixers into three component functions:

1. The method for deciding when to raise the gain of an input channel
2. The dynamic action of the input channels (gate envelope)
3. The method for number-of-open-mikes (NOM) gain adjustment

Each of these points will be discussed in order.

Methods for Deciding When to Raise the Gain

Building on the knowledge that the VOX approach was inadequate, several patented methods for detecting an active input have been successful in automatic mike mixers. They are:

a. To use one or more ambience mikes

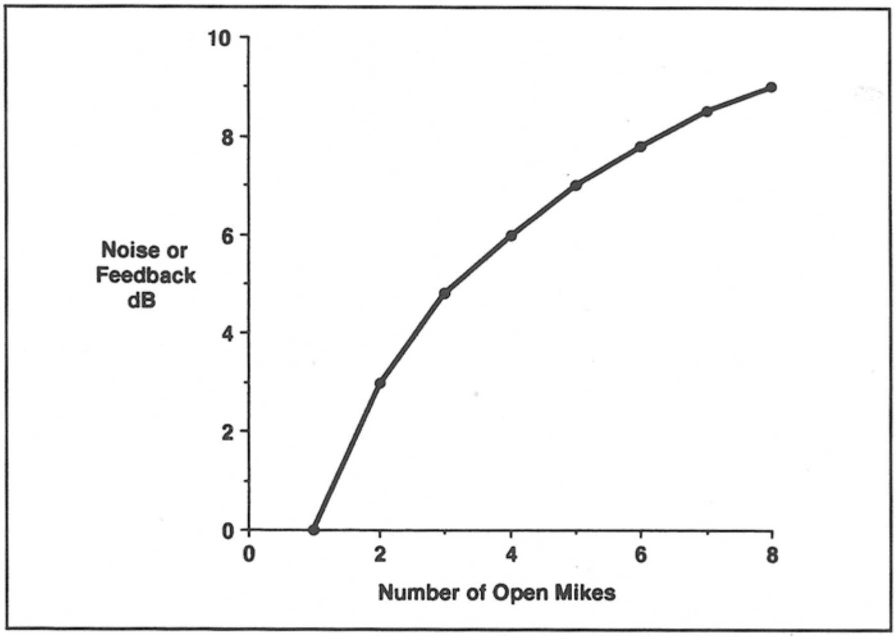

Figure 1. The increase in feedback or ambient noise as more mikes are opened. It is assumed that all the mikes have the same acoustic gain. The function is 10 log NOM.

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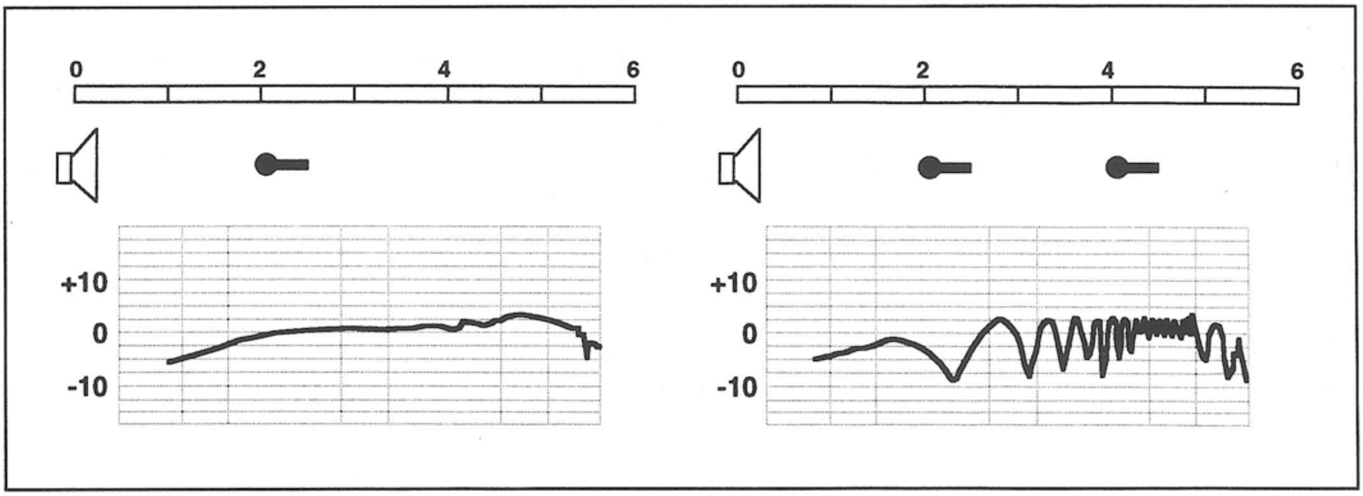

Figure 2. When two microphones are open at the same gain, the time-delayed sound picked up by the second mike creates a comb-filter effect that degrades the tone quality.[1]

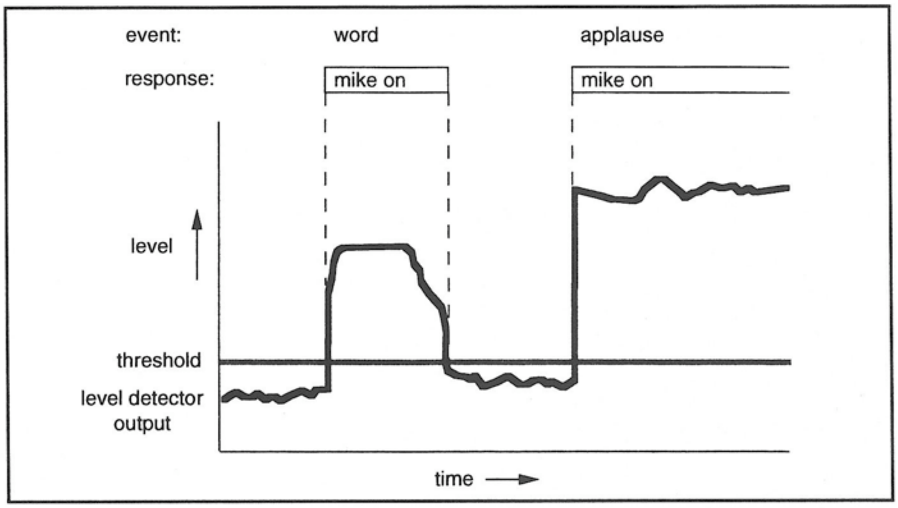

Figure 3. The reactions of a fixed-threshold gate or VOX to two acoustic events, one desired and one not.

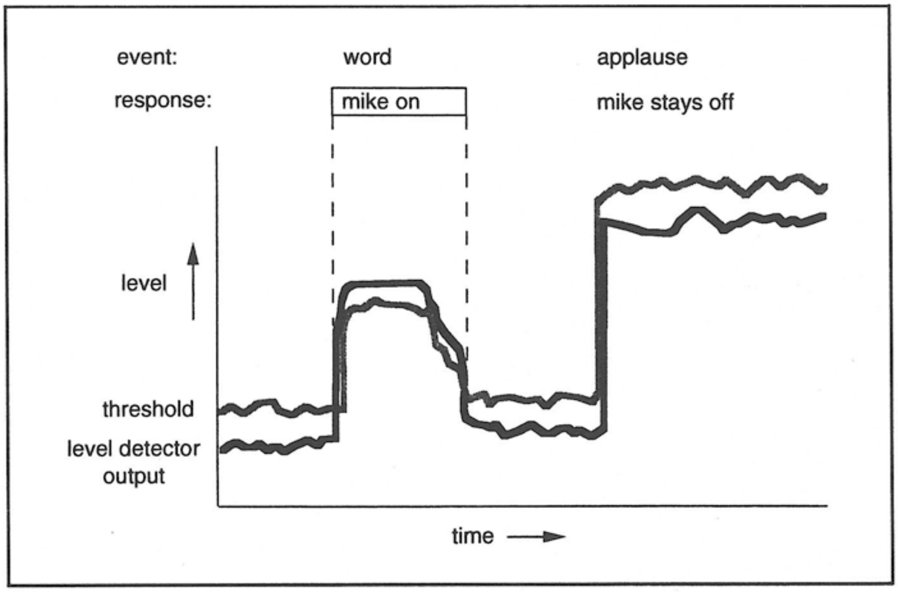

Figure 4. The same two events as shown in Fig. 3, showing how an adaptive threshold system avoids actuation even by a high level of ambient sound.

to create an adaptive threshold or individual adaptive thresholds for the input channels[3,4] (Figs. 4, 5)

b. To normalize the pattern of input levels and transform it continuously into a pattern of input gains[5] (Fig. 6)

c. To select the highest level input at any moment and hold it on for a short, fixed time[6] (Fig. 7)

d. To use the sum of the levels of the active inputs to vary the thresholds[7] (Fig. 8)

The Dynamic Action of the Input Channels

Those automatic mixers that use gating, rather than expansion or gain shifting, impose a characteristic time envelope on the signal being gated (Fig. 9). The GATE ON command is followed by the input switching on, or, better, a fixed-rate gain ramp up, typically 3 to 15 msec. The gate is held open as long as the command continues. When the command ends, there may be a hold time intended to keep the gate open during the short pauses in running speech, typically 0.2 to 0.4 sec. After the hold time (if any), the gain is either switched to its low level, or, in gentler systems, ramped down, typically over 0.3 sec.

Careful attention to attack, hold, and decay times can do much to approach a tolerable sound in an automatic mixer. Mixers having an expansion function rather than a simple gate produce a much better sound when they are operating close to the threshold (Fig. 10). With an expander, the attack and decay of the gate is controlled by the rise and fall rates of the signal level itself.

Methods of NOM Adjustment

Most automatic mike mixers use a master attenuator to perform the number-of-open-microphones gain adjustment. In sound reinforcement this function prevents feedback. In recording or broadcasting it prevents the build-up of ambient noise as additional mikes are opened up simultaneously (Figs. 11, 12).

There is no master NOM attenuator in the Dugan speech system mixers (Fig. 6) because the NOM function is implicit in the gain shifting function. The combination of the gains of the input channels under any condition of stimulation is equal to the gain of one open microphone.

A recent development by the author improves on the NOM attenuator.[8] Instead of counting the number of open microphones, it keeps track of the total gain of all the channels and reduces the master gain continuously and proportionately whenever a "gain limiting" threshold is exceeded. This system takes into account the build-up of gain from "off" channels (typically down 15 to 20 dB) and is intended for more demanding or artistic applications. The gain limiting function appears in the prototype described below.

Extending the Field of Automatic Mixer Applications

Automatic mixers have been in use in permanently installed sound-reinforcement systems for 15 years. At the last count versions were being produced by nine U.S. manufacturers and one in Japan. Though they differ in their working principles and in the various remote control and logic output functions they offer, they are all the same in that they are all simple mixers intended for unattended operation. The applications in which they are often found include churches, courtrooms, civic council and board chambers, corporate boardrooms, legislatures, teleconferencing, and radio talk studios.

It is now possible to apply the better-sounding automatic mixing systems to human-operated systems. Many multimike mixing operations can be simplified, and many common mixing errors can be eliminated, by having automatic mixing functions available to the operator. The untapped applications include broadcast news panels, broadcast talk and game shows, motion-picture location recording, conference sound reinforcement, conference video production, churches with mixing consoles, board rooms with mixing consoles, Broadway shows, casino showrooms, opera houses, and rock concerts.

From a human engineering standpoint it would be best if automatic mixing were integrated into the console design, so that the controls would be right at hand and the gain structure fully integrated with the console. At this stage of development some economic considerations dominate. Consoles are not replaced very often, and prototyping a mixing console is very expensive. In light of these facts, the decision was made to start with an external proces-

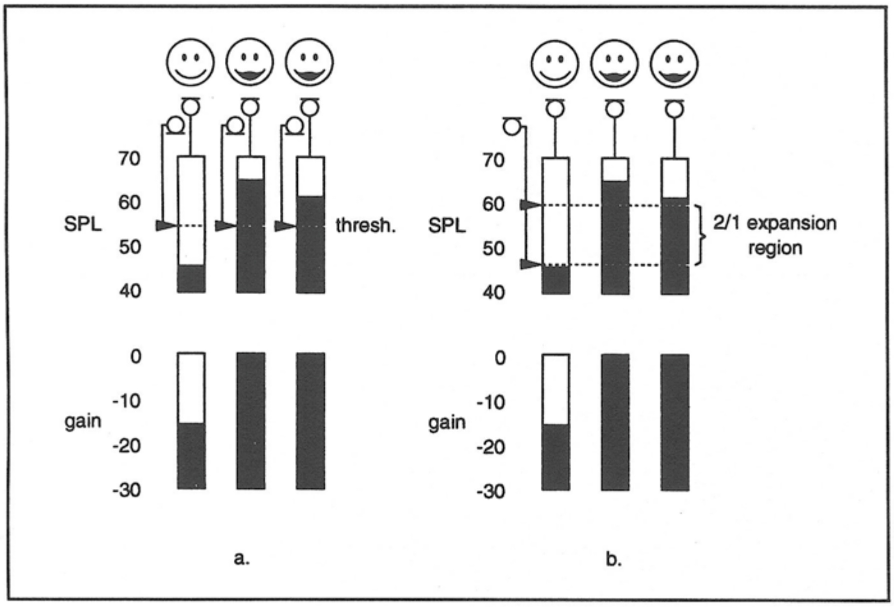

Figure 5. Two adaptive threshold schemes using separate ambience sensing: (a) using a mike to set a threshold for each input;[4] (b) using a single mike to set a threshold for the inputs (Dugan music system, see Fig. 10).[3]

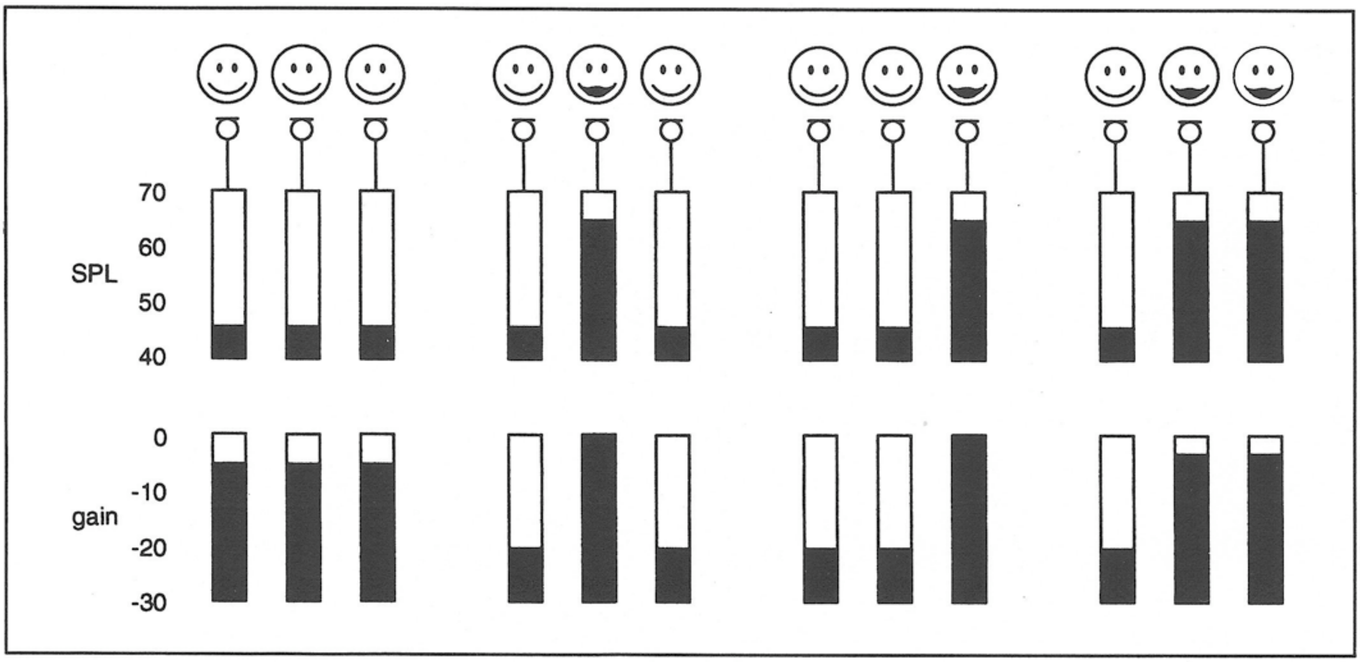

Figure 6. The action of the Dugan speech system. The pattern of levels at the inputs is turned into a pattern of gains. The absolute level is ignored, so that the talker's level may vary while the gain structure will remain centered on his mike.

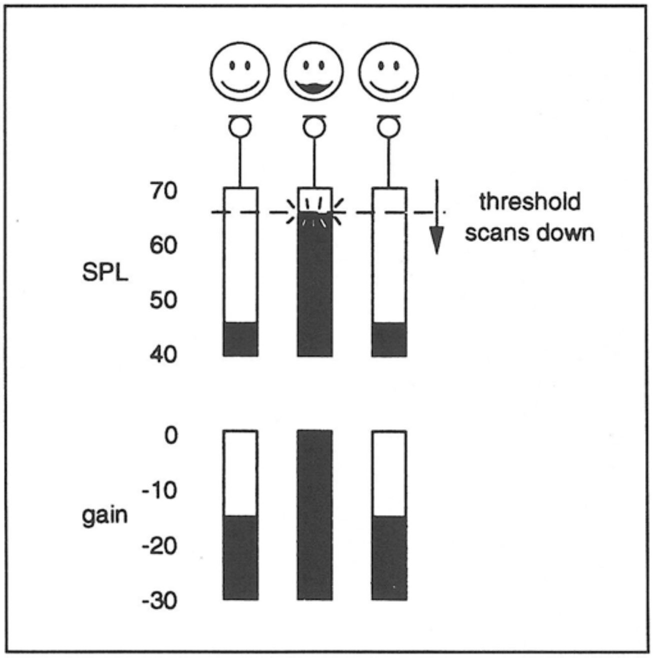

Figure 7. In the Peters patent,[6] a threshold scan starts at the top and scans rapidly downward. It resets for a new scan whenever it hits the highest level input or the bottom of its range. The mike "hit" is turned on for a short hold time.

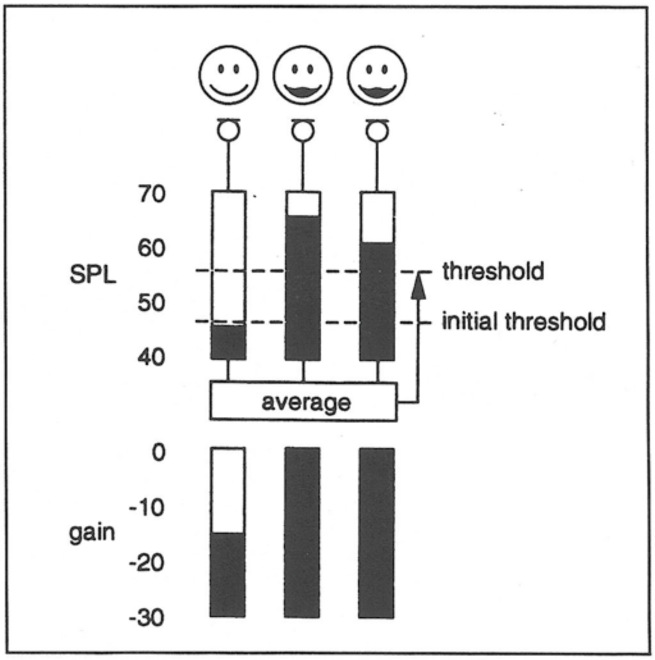

Figure 8. The action of the Ponto and Martin patent[7] and several other schemes. A low initial gating threshold is modified by the average of the input levels, using the sum of the levels of the active inputs to vary the thresholds.

sor that can be patched into any console. This format would always be useful with older equipment, even if automatic mixing functions were to become common in new consoles.

A Design for an Automatic Mixing Controller

The prototype automatic mixing controller presented here consists of eight discrete audio voltage-controlled amplifier (VCA) channels with an analog control system. The control system offers five functions, which can be used alone or in combinations. These are the Dugan speech system, the Dugan music system, gain limiting, last hold, and level servo.

All these functions except the level servo are methods of automatic microphone mixing, automating the routine task of fading mikes in and out as they are needed. Active channel signals are passed through the controller at unity gain with no change in dynamics or frequency response. When the Dugan speech system is used, it will automatically and silently cross-fade mikes to maintain constant ambience pickup in broadcast and recording. When the Dugan music system is selected, the controller will instantly fade any active mike up to full gain, and fade unused mikes down, for optimum sound reinforcement without feedback. The last hold function may be used to provide a constant ambience background in conjunction with the music system. Gain limiting is used along with the music system to prevent feedback or noise buildup as more mikes are brought into use. If automatic level control is desired in addition to the automatic mixing, the level servo may be switched on. The level servo maintains a constant volume unit (VU) level in each channel and holds its gain setting when the input is not in use.

The controller is a line-level device with unity gain. The inputs are active balanced and the outputs are low-impedance (50 Ω actual) unbalanced, capable of driving 600-Ω loads. It is intended to work in the insert points of a mixing console, after the mike preamps (Fig. 13). The audio path is kept very short and simple, consisting only of an input buffer, a high-quality VCA, and an output buffer (Fig. 14). The functions controlling the VCA gain will be described in detail.

The Dugan Speech System

This speech system has been applied successfully in installed sound-reinforce-

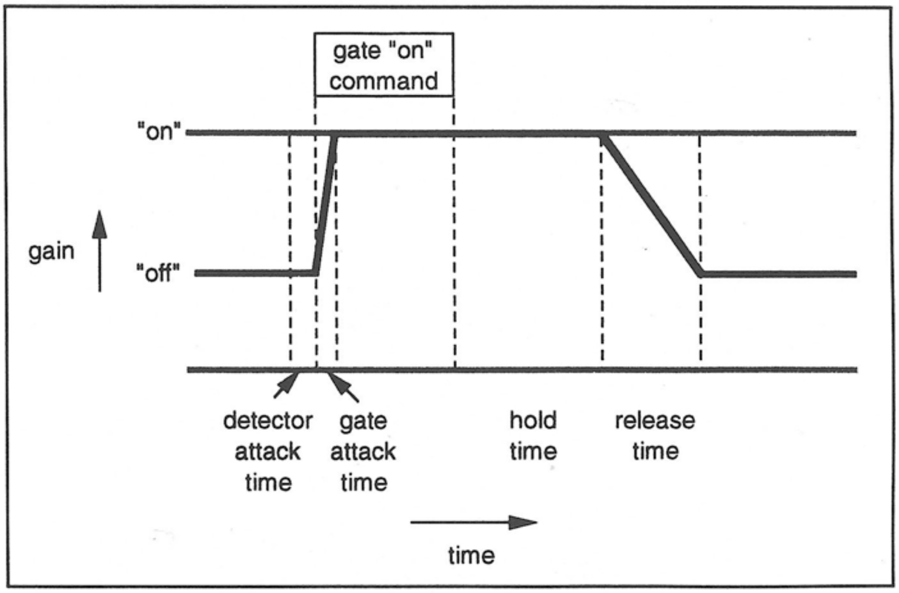

Figure 9. The terminology of gates. The hold time function keeps the gate open for a fixed time after the stimulus has fallen below threshold. Some call the hold time "release time." In a gating mixer the hold time is important to bridge between words and prevent the chopping off of syllables. A few products have fade-out release times. In an expander gate, the attack and release times are determined by the level detector's timing.

ment systems for many years. This function is very easy to use because there is no need for any special equipment except the automatic mixer itself, or for threshold adjustments. It computes the ratio of each input's level to the level of the sum of all the inputs, and makes the gain of the input equal to that result. It is a continuous function without thresholds or gating.

As depicted in Fig. 6, when no one is speaking, all the input channels will hover at an intermediate gain level. The sum of the gains of all the inputs will equal the gain of one microphone at normal gain. This provides a natural-sounding amount of room ambience in the mix. When someone speaks, his/her channel's gain fades up rapidly (approx. 15 msec) to normal gain, and all the other channels are pushed down to a low gain. As long as the speaker continues, there is no change in the dynamics of the channel, the channel remaining at normal "on" gain. In case of interruptions or arguments, the system will unobtrusively share the gain among the competing microphones in proportion to their relative levels.

When this function is being used, all the auto mix threshold controls (Fig. 15) are usually left at the "0" mark. The threshold controls may be used to fine-tune the function in certain situations. If, for example, the input gain trimmers on the console have been set so that the ambient noise levels in the channels are quite different, these differences may be compensated for on the controller's threshold controls without altering the gain structure as set up on the console. This is easily done by observing the behavior of the auto mix gain meters during a pause, and adjusting the threshold controls so that all the channels have approximately equal gain reduction during the ambient noise condition. In the speech system the auto mix threshold controls are not actually adjusting thresholds (in this function there are none), but the relative weightings of the input channels in the control system.

Another case where the auto mix threshold controls would be useful with the speech system would be where one mike happens to be in a particularly noisy location, perhaps near an air vent. That channel may be deemphasized in the control system by lowering its threshold control. This will prevent the noise from that area from dominating in the ambience mix, without changing the gain of that mike in the mix when it is in use.

The Dugan Music System

The speech system is ideal for mixing essentially one-at-a-time sources; it expands (downward, 2× in dB) the instantaneous level differences between mikes. In music mixing, however, this is not desirable. In music, the relative levels of simultaneous inputs must be kept unchanged; therefore a different function is called for.

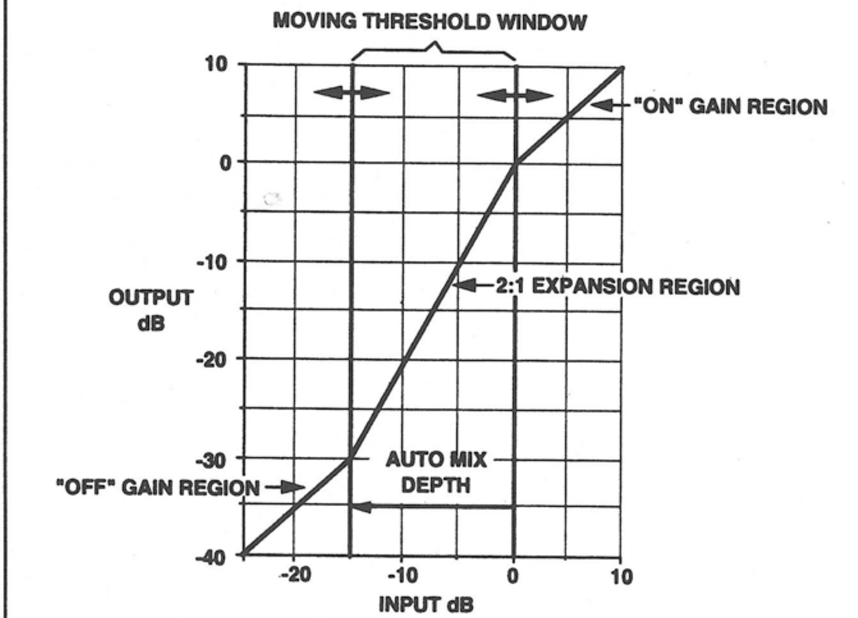

Figure 10. The input-output level function in an adaptive-threshold gate with expander action. The whole 2:1 region slides from left to right depending on the varying threshold (Fig. 5b).

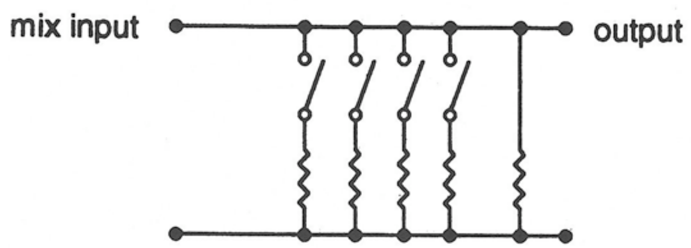

Figure 11. This simple NOM attenuator designed by Ken Patterson around 1968, or a mix-bus loading variant of the same idea, is in the public domain and is used in many automatic mixers. As each microphone opens, one switch is closed. Surprisingly, it approximates the 10 log NOM curve fairly closely for the first few mikes.

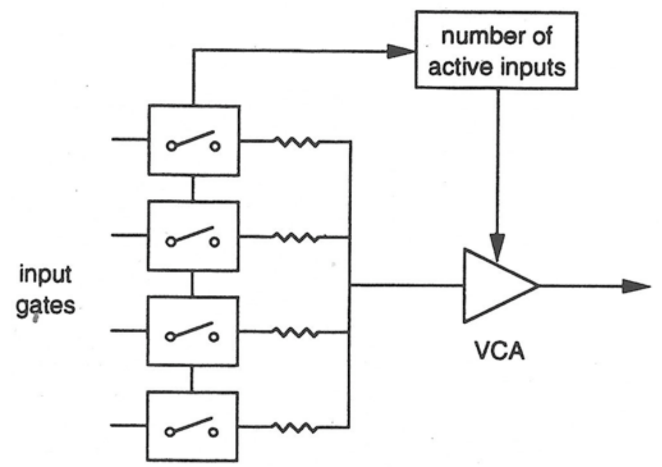

Figure 12. Those automatic mixers not using the Patterson attenuator use some variation of the above. A master VCA or stepped attenuator is controlled by some means for counting the number of mikes that are open at the moment.

In sound reinforcement it is highly desirable to mute mikes that are not being used, or to pull down their faders enough so that they do not contribute confusion and feedback to the mix. Mixing-console operators do this "defensive mixing" all the time; in fact, it is this activity that usually constitutes the majority of their moves. Automatic muting or pull-down would be very helpful to live-music mixers, but little help is available in current technology. Studio-type expander gates are sometimes seen applied to multiple drum-set mikes on stage, where they are viable because the close-miked drums are very loud at the mikes, and a fixed-threshold VOX (Fig. 3) can be made to work.

Gating is rarely practical with vocal and instrumental sources because of their dynamic range. If the gate on a vocal mike were set to keep it off during loud instrumental passages, it would only respond to loud vocals and act dead or intermittent during a softer song. Conversely, if an expander gate were set to open for soft vocals on a quiet stage, it would be open all of the time during the loud parts. Therefore gates are of little use to the music mixer.

The solution to this problem is to use adaptive threshold expander gates (Figs. 4 and 5). One adaptive threshold gating system uses special dual-element mikes that derive an acoustical signal-to-noise ratio at each mike location.[4] The requirement for special mikes makes music applications of this method problematic, as the selection of microphone types is very important in music mixing.

The Dugan music system[3] (Fig. 5b) uses one ambience mike which, if carefully placed, can provide the threshold reference for the whole stage system. The reference mike is patched from the direct output of a console input strip to the music system threshold input on the automatic mixing controller. In the controller, a 2:1 ratio downward expander on each input has a continuously varying threshold that depends on the sound level sampled by the reference mike. The auto mix threshold control on each input (Fig. 15) is set so that the input comes to normal (or "on") gain when the source is far enough above the general stage sound level to be useful, typically at least 10 dB. As long as the input is significantly above the confusion — that is, as long as the input is mixable — there is no change in its gain, and both loud and soft playing or singing are

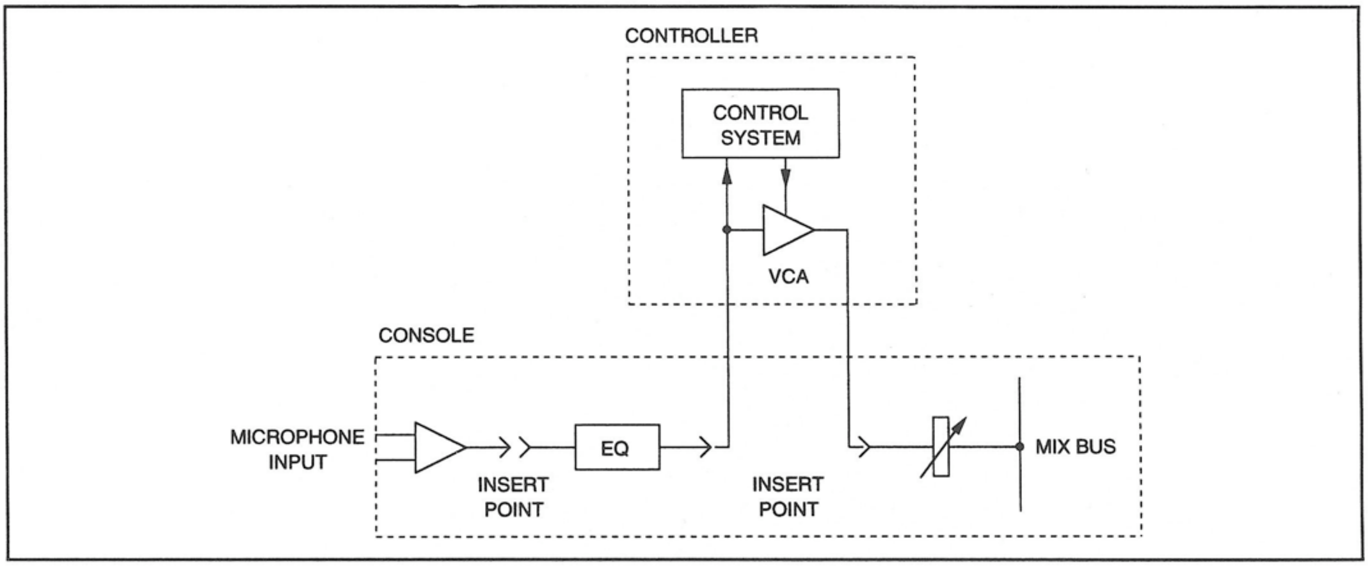

Figure 13. The automatic mixing controller is connected in the insert points of console input channels.

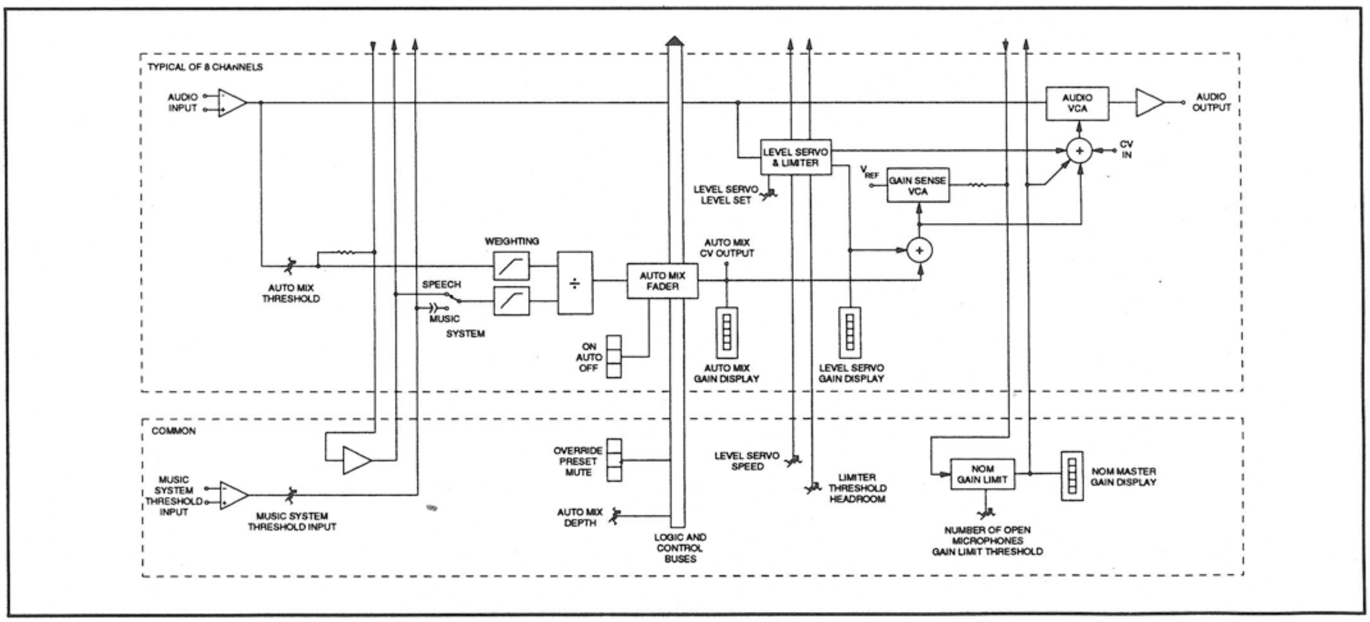

Figure 14. Block diagram of the automatic mixing controller.

faithfully transmitted without dynamic alteration. When the stage is quiet, the channels become very sensitive and distant mike techniques may be used.

In some situations, a single reference mike will not serve. In an opera house, for example, one reference mike could be used for area and wireless mikes on stage, sampling the general stage ambience, but a mike for an offstage singer might be subject to a quite different ambience. A separate reference mike should be provided for the offstage effect. The controller includes patch points at which separate music system threshold signals may be inserted in each channel.

The gain of the "off" or faded-down inputs may be set by the auto mix depth master control (Fig. 15). Subtle effects such as pulling the inputs down only a few decibels when they are not in use may be achieved.

Gain Limiting

As more mikes are brought into a sound reinforcement mix the feedback from the loudspeakers is increased. A mix of eight mikes must be pulled down about 9 dB compared to the gain achievable with any one of them alone (Fig. 1). Most automatic mixers available for installed sound systems count the number of open mikes and apply the 10 log NOM curve to the mix. This tactic assumes that all the mikes are operated near feedback, and that the "off" attenuation is enough to make the contribution of attenuated channels insignificant. Neither of these assumptions may hold in a performance or broadcast sound system.

The new "gain limiting" function[8] provided in the automatic mixing controller is sensitive to the gain contribution of "off" channels. A running sum is kept of the gains of all the input channels that are being automatically mixed. These gains vary between normal "on" gain and various amounts of attenuation depending on the instantaneous state of the automatic mixing and the automatic level servo (see below). Whenever this sum exceeds the allowed gain set by the gain limit threshold control (typically NOM = 1), the whole mix is attenuated just enough to keep the overall system below feedback. Since there is no master audio channel in the controller, this is done by attenuating all the channels by the same amount. Note that the gain limit has nothing to do with the signal levels, only with the gain conditions of the respective channels as determined by the automatic mixing functions and the automatic level control.

In the common situation where the acoustical gain is marginal, where pushing up the fader runs into feedback before a satisfying volume is reached, an NOM gain limit threshold of 1 is appropriate. When one mike is in use, it can be mixed at the maximum possible gain without fear that the entry of other voices will cause the system to howl. When a second voice joins in there will be an automatic master gain pull-down of 3 dB. A third voice will cause an additional reduction of 1.8 dB, and a fourth voice will reduce the gain about 1.2 dB more. The transition from one mike to two is the largest step, and that amount of gain reduction is easily masked by the entry of the new sound into the mix. It is inaudible unless consciously listened for.

In a system where there is some leeway for gain, but feedback will become a problem when a greater number of mikes are in use, another strategy may be used. The operator may start with the gain limit threshold set at a high number, and then when feedback threatens, reduce the threshold just to the point where sufficient gain reduction occurs to keep the system safe. Now when a lesser number of mikes than the threshold setting are in use, there will be no master gain adjustment. The adjustment will start gradually when more mikes come up and the gain limit threshold is exceeded (Fig. 16).

In the preceding two paragraphs, mike inputs have been talked about as "in use," meaning at normal "on" gain, or "off," meaning attenuated. Since the gain-limiting function is monitoring the actual gain, not just on or off, it responds continuously to all in-between gain conditions too. For example, if at a given moment the automatic mixing and/or level servo functions result in one mike being fully on, and four other inputs partly attenuated at -6 dB, the resulting master gain adjustment would be -3 dB. This accounts for the four mikes at -6 being equal to one at 0, which combined with the first at 0 equals an NOM of two. When the system is in gain limiting and a source stops, that input is attenuated and the available system gain is "given back" to the mix.

Since NOM gain adjustments are made to all channels together, the action of the gain-limiting function does not alter the musical balance. Nothing changes the balance of voices the operator has made with the faders on the console. The automatic mixing functions do not produce pumping or breathing effects either. After channels have come up to normal gain, their gain is constant.

Where feedback is not a problem, for example in broadcasting or recording, there is another use for gain limiting. It

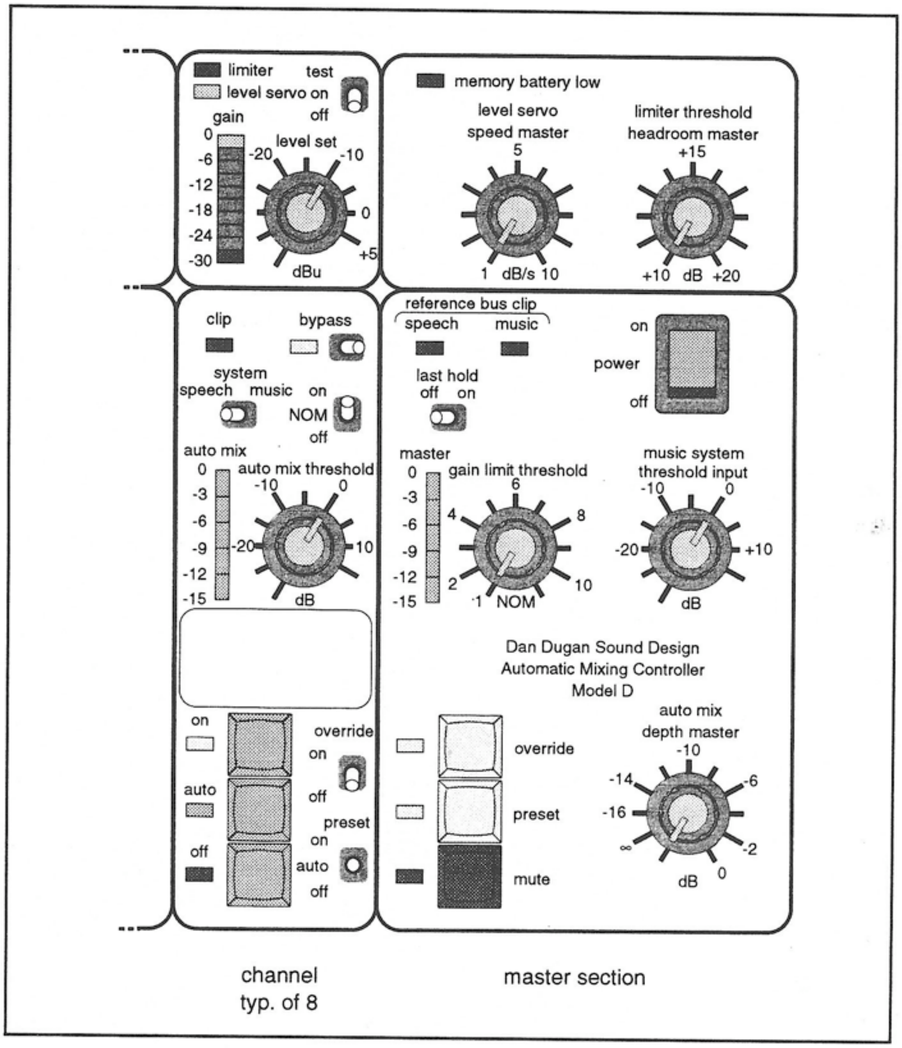

Figure 15. Control panel of the automatic mixing controller.

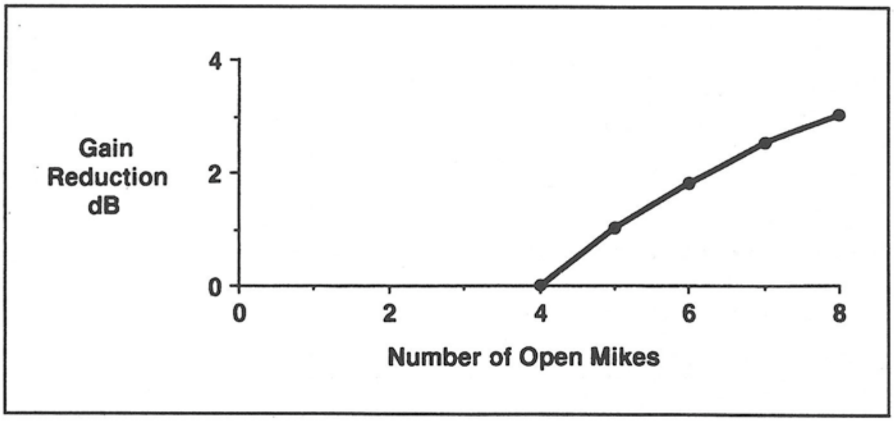

Figure 16. The action of the gain limiting function when the gain limit threshold is set to four mikes (compare with Fig. 1).

may be used to regulate the room or environmental noise, so that when more mikes are in use the ambient noise does not build up.

The Last-Hold Function

The Dugan speech system maintains a "one mike" ambience pickup during program pauses, so with the speech function a natural ambience effect is transmitted. When the Dugan music system is being used, the ambience in the mix will be suppressed. This is desirable in some situations, but for recording or broadcasting the effect may be unnatural, with room noise disappearing during pauses. One remedy for this is logic circuitry, which holds the latest microphone in use on during program pauses. This function is found in several current automatic mixers. In the automatic mixing controller, the last-hold function is available under the control of a front-panel switch (Fig. 15).

A Digital Memory Automatic Level Control

Automatic level control (ALC) is desirable where input levels may vary or where hands-off mixing is desired. Simple compressors or ALCs are not very satisfactory because they pump up the gain to the maximum during program pauses. This shortcoming led to the development of "gated" level controls that hold the gain when the signal falls below a threshold. Note that the term "gating" in this context refers to holding or releasing the level control gain, not to attenuating the program signal. Gated automatic level controls use the signal level to determine when the ALC should be allowed to release.

In an automatic mixer very good intelligence is available as to whether the input is in use or not. The automatic mixing controller described here has a level control system with two novel features.[9] First, level memory gating is determined by the automatic mixing functions. Whenever a channel's auto mix gain is above -3 dB, the logic output for that channel is true, and the level servo system is enabled for adjustment. When the level servo is being used without automatic mixing, i.e., when the channel is in the "on" (nonautomatic) mode, the level servo reverts to the conventional ALC gating technique, enabling the servo whenever the signal level is above a threshold. This threshold is 15 dB below the desired output level set on the level servo level set control.

A second unique feature of the ALC provided in the controller is permanent memory. Previous embodiments of the gated ALC idea have drifted slowly to a preset "idle gain" after some time without activity in the channel. This ALC system keeps its gain setting in a low-power memory even when the power is turned off. In a church, for example, when the system is turned on again after a week off, the channel gains will be the same. This is why it is called "level servo." The action resembles a motorized fader that arrives at the desired level and stays there indefinitely.

When the channel is active, the level servo system looks at the channel's output level through a filter simulating the response of a VU meter. This is compared to the desired output level that the operator has set on the level servo level set control. If the level isn't close to the target, the level servo will slowly fade the gain up or down until it matches. In program pauses it freezes where it is.

The automatic gain-riding speed is set by the level servo speed master control (Fig. 15), which has a range from a slow sneak to something more like a compressor. An operator might set the speed faster at first to get quick levels for all the mikes, and then turn the speed down to slow to make the system more immune to sudden noises, such as when someone hits a mike accidentally.

Protection against sudden level increases is provided by a fast peak limiter. The limiter threshold is set by the limiter threshold headroom master control, with which the threshold can be positioned from 10 to 20 dB above the level servo level set. A sudden loud sound will actuate the limiter, which releases quickly so that interrupting sounds won't "punch holes" in the program. A sustained excessive level will continue to activate the limiter for a few seconds until the slower level servo catches up. Thus the limiter only operates briefly when it is needed.

Automatic level control must be used with great care in sound reinforcement. If the other channel gain controls (e.g., console trimmer or fader) are raised while an ALC is in gain reduction, the system may feed back later when the ALC comes up to a higher gain. To aid the operator, the level servo on/off switch has a TEST position (Fig. 15). This spring-return position brings both the channel automatic mixing and the level servo to full gain while the switch is held, without losing the level servo memory gain setting. The operator should use this switch to check that the channel is, ideally, just below feedback in this maximum-gain position. When TEST is released the channel goes back to its automatic mixing mode (if any) and level servo condition. TEST should be used when adjusting the console input trimmer and fader. The level servo gain memory may be reset to 0 dB (full up) by momentarily switching to OFF.

Other Features and Remote Control

Automatic Fading

The automatic mixing functions may be

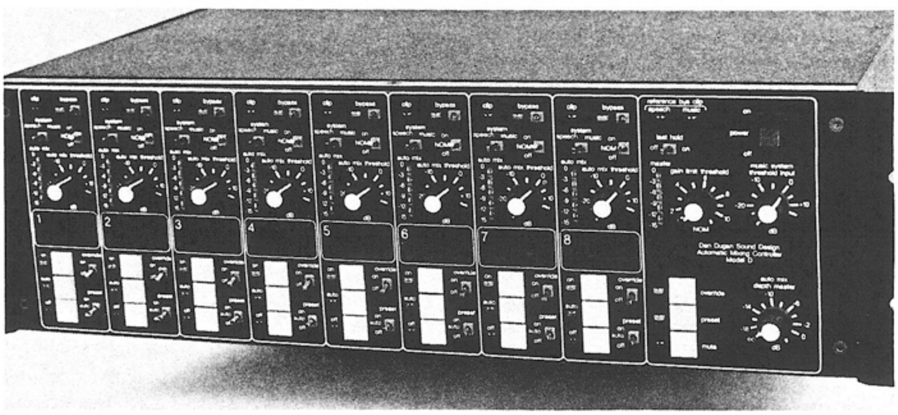

Figure 17. Pilot production model of the automatic mixing controller.

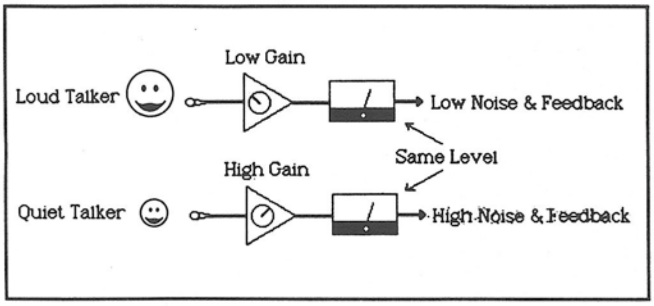

Figure A1. Gains set to match the levels of two talkers; matching levels results in different acoustic gains.

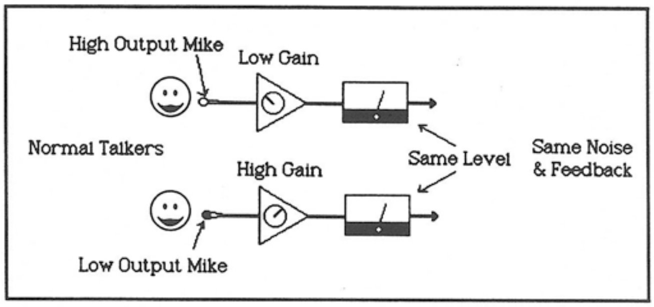

Figure A2. Gains adjusted to match different mikes; different electrical gains result in the same acoustical gain.

switched in or out of action by means of ON, AUTO, and OFF keys (Fig. 15). When the channel's mode is changed by these keys, the resulting gain change is made as a quick-fade of about ½ sec. For example, if a change from ON to AUTO would produce a gain change of -10 dB, hitting AUTO will make a quick fade down to -10. This feature may be used to make smooth cue-in and cue-out fades to the level preset on the console by hitting ON or OFF. The settings of these modes at power-up can be programmed by the PRESET switches on each channel (see below).

The Override Controls

At times it is desirable to override the automatic mixing system and bring the channels to a preset combination of on and muted conditions, for example, to bring up the chairperson's mike and mute all others. Each channel has an override assign switch (Fig. 15) that determines what the channel will do when the OVERRIDE key is pressed. Channels assigned OVERRIDE ON will quick-fade up to full gain, and channels assigned OVERRIDE OFF will quick-fade to full mute.

Preset Key

During a discussion, the operator may have some mikes muted, and suddenly become aware that someone is talking to a dead mike. The PRESET key provides a panic button for this situation. It puts all channels into the mode (ON, AUTO, or OFF) that the operator has programmed with the PRESET switches on each channel.

Hard-Wire Bypass

Each channel has a bypass switch controlling a relay that connects the input to the output. When the unit is not powered all channels are bypassed, so the console channels operate normally.

Remote Control Terminals

It is advantageous if the automatic mixing controls, particularly the ON/AUTO/OFF keys and the auto mix gain meter, are positioned near the channel's fader on the console. The controller provides terminals to make such an installation easy. Computer control should be simple to implement also. The internal high-quality VCA may be controlled by an external voltage source. Gain changing from outside will not affect the automatic mixing system. The level servo and limiter will act on the channel output level resulting from both internal and external gain changes.

All logic controls are active low, so that only a contact closure to a common ground is needed. The following connections are provided for each channel.

Inputs:
- SET ON
- SET AUTO
- SET MUTE
- CHANNEL UP (momentary)
- CHANNEL MUTE (momentary)
- VCA Control Voltage

Outputs:
- ON LED
- AUTO LED
- OFF LED
- Logic Out (channel auto mix gain above -3 dB)
- Control Voltage Output (for a remote auto mix gain meter)

The following terminals are provided for the general system controls.

Inputs:
- OVERRIDE (momentary)
- PRESET
- ALL MUTE (momentary)

Outputs:
- OVERRIDE LED
- PRESET LED
- ALL MUTE LED

Multiple Unit Linking

A link connector is provided for combining the functions of multiple automatic mixing controller units. The system is designed to handle linked systems of up to 100 inputs. Units are daisy-chained with special cables. All the automatic mixing functions, including gain limiting and last hold, are interactively linked, and so are the master control keys OVERRIDE, ALL AUTO, and MUTE. The auto mix depth is local on the eight-channel units. The level servo speed and limiter threshold headroom remain local on each unit. The level servo action is independent in each channel.

Pilot Production

Figure 17 is a photograph of a pilot production model implementing all of the aforementioned functions except the level servo. This model includes jacks for connecting an add-on ALC unit.

Conclusion

This article has presented some theoretical background on automatic mixing and a proposal for hardware that is expected to improve the quality of live audio production. The author hopes that his readers are now better prepared to discriminate between simple on-off automatic mike mixing and more sophisticated functions. Field experience with the pilot production of the controller will test the worth of these ideas and help shape future developments.

Appendix

Understanding Gain and Level

Level refers to the sound pressure level or the audio voltage in the system at a given moment. It varies depending on what is happening at the microphone. There will be an ambient noise level, and a program level when someone's talking.

Gain is the amount of amplification in the channel between the input and the output. It changes when the mixer controls are changed. However, any time the system gain is changed the levels will change, and the only way to control the program levels is by changing the gain applied to the signal (Figs. A1, A2).

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