Automatic Microphone Mixer

White Paper

April 2013

Abstract

Automatic microphone mixers are useful for mixing engineers to manage multiple live mics without riding individual input faders, especially in unscripted programs. This document explains the basic benefits, the acoustic gain structure, the difference between gating type and gain sharing type, and the Dugan-MY16 as a product example.

1. What Are Automatic Microphone Mixers?

Automatic microphone mixers (or automixers) are useful in situations where multiple mics are used, such as in conferences, panel discussions, stockholders' meetings, and recording of talk shows. They can alleviate the mixing engineer's complicated fader operation especially in unscripted programs.

Automatic microphone mixers solve the following three problems that occur when multiple mics are used.

➢ Reduction in feedback margin

When multiple mics are open, the feedback margin is reduced. Each time the number of open mics doubles, the feedback margin decreases by 3 dB. Compared to when there is only one mic, the margin decreases by 3 dB when there are two mics, by 6 dB when there are four, and by 9 dB when there are eight.

➢ Increase in amplified ambient noise

Let's consider an eight-member meeting where a mic is placed in front of each member. If all eight mics are open even though there is only one member speaking at any given time, the eight mics will pick up and amplify the ambient noise reducing the SN ratio of the target speaker's amplified voice to the amplified ambient noise.

When eight mics are open, the amplified ambient noise will be 9 dB higher than when only one mic is open. In other words, the SN ratio is reduced by 9 dB degrading the clarity of the voice.

➢ Generation of comb filters

Comb filters are generated from overlapping mic signals. For example, if a sound source is picked up by two mics, the difference in the distances from the sound source to each mic creates a time difference; in other words, a phase difference. When these two mic signals are mixed together, a comb-shaped filter is generated and reduces the sound quality. Comb filters that are generated at frequencies 1 kHz and higher greatly affect voice clarity.

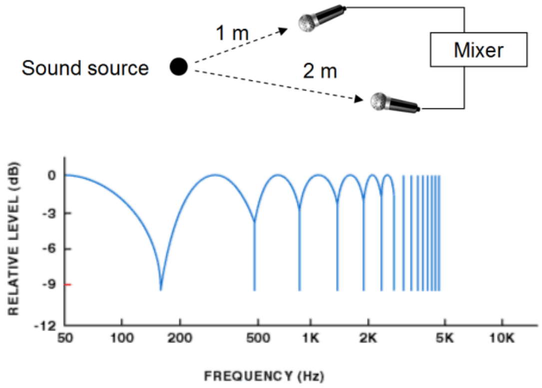

Figure 1

Automatic microphone mixers automatically reduce or turn off the gains of mics except for the mic that is receiving input. If you have a system where feedback does not occur using one open mic, an automatic microphone mixer can automatically adjust the acoustic system gain to prevent feedback even when multiple mics are open. These features solve the three problems mentioned above.

2. Acoustic Gain

To fully understand automatic microphone mixers, you must understand the concept of acoustic gain.

To amplify voice, you need acoustic gain. Acoustic gain is defined as the increased sound pressure level (gain) that the listener receives when a sound reinforcement system is in operation as compared to when the system is not in operation.

A normal conversation in a quiet room is 65 dBSPL at a distance of 1 m. If the farthest listener is 10 m away, the sound pressure level at that distance is 65 - 20log(10) = 45 dBSPL, which is 20 dBSPL less than the sound pressure level at 1 m away.

The purpose of a sound reinforcement system is to amplify the voice picked up by a mic to give an impression that the speaker is closer to the listener. In order to give an impression that the speaker 10 m away is only 1 m away from the listener (to achieve a sound pressure level of 65 dBSPL at 10 m), the reinforcement system must have a gain of at least 20 dB.

This gain is called acoustic gain or needed acoustic gain (NAG). The acoustic gain varies depending on the distance between the speaker (the speaking person), mic, loudspeaker, and listener*1. Details are explained below using the following figure.

D_1: Distance between the mic and the loudspeaker
D_0: Distance between the speaker and the farthest listener
D_2: Distance from the loudspeaker to the farthest listener
D_S: Distance from the speaker to the mic

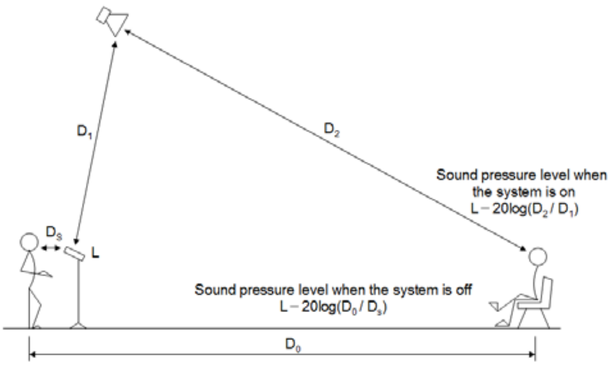

Figure 2

- When the system is inactive

If the sound pressure level applied by the speaker to the mic is represented by L, from the inverse square law, the sound pressure level at the farthest listener is expressed by the following equation.

Sound pressure level at the listener position farthest from the speaker

$$= L - 20\log(D_0/D_S) \quad \text{................} \quad \text{(Eq. 1)}$$

- When the system is active

Here, we will assume that the mic and loudspeaker are omni-directional. We increase the system gain to a setting just short of the setting that would cause a feedback. At this point, the gain between the mic and loudspeaker is 0 dB (unity gain). (If feedback occurs, the gain is positive. If not, the gain is negative.) This condition indicates that the speaker and loudspeaker sound pressure levels, individually captured by the mic, are equal. We can then calculate the resulting sound pressure level of the loudspeaker at the position of farthest listener.

Sound pressure level at the listener position farthest from the speaker

$$= L - 20\log(D_2/D_1) \quad \text{................} \quad \text{(Eq. 2)}$$

The acoustic gain is defined as the difference between Eq. 1 and Eq. 2, so it is expressed by the following equation.

Acoustic gain

$$= (L - 20\log(D_2/D_1)) - (L - 20\log(D_0/D_S))$$
$$= 20\log(D_0/D_S) - 20\log(D_2/D_1)$$
$$= 20\log(D_0/D_S) + 20\log(D_1/D_2)$$
$$= 20\log D_1 + 20\log D_0 - 20\log D_2 - 20\log D_S \quad \text{.....} \quad \text{(Eq. 3)}$$

The acoustic gain in Eq. 3 is the acoustic gain at the feedback threshold level. An acoustic system cannot be run in this condition, so a margin of safety (feedback margin) is required. Normally, the feedback margin is at least 6 dB. The maximum acoustic gain that an acoustic system can produce, which includes the feedback margin, is called the potential acoustic gain (PAG).

Potential acoustic gain

$$= 20\log D_1 + 20\log D_0 - 20\log D_2 - 20\log D_S - 6 \quad \text{...} \quad \text{(Eq. 4)}$$

As an example, let's substitute the following values into the variables: $D_1 = 5$ m, $D_0 = 16$ m, $D_2 = 13$ m, $D_S = 0.5$ m in Eq. 4.

Potential acoustic gain

$$= 20\log 5 + 20\log 16 - 20\log 13 - 20\log 0.5 - 6 = 15.8 \text{ dB}$$

In the above description, the mic and loudspeaker are assumed to be omni-directional. In an actual system, a directional mic and loudspeaker are used, so the potential acoustic gain can be increased. The main point here is that automatic microphone mixers operate to keep the potential acoustic gain constant at all times.

3. Effects of Multiple Open Mics

Up to this point, the acoustic gain in a system was considered to use only one mic. In this section, a system using multiple mics will be considered.

If two mics with the same characteristics are set to the same gain, the total system gain increases by 3 dB. To suppress feedback, the total system gain must be decreased by 3 dB.

Each time the number of open mics (NOM) doubles, the system gain increases by 3 dB. So if there are four mics, the system gain increases by 6 dB. If there are eight, the gain increases by 9 dB. If we express this as an equation, we get Eq. 5:

NOM (dB)
= 10logNOM where NOM is the number of open mics. ... (Eq. 5)

If we incorporate the NOM variable in the potential acoustic gain equation, we get the following:

Potential acoustic gain
$= 20\log D_1 - 20\log D_2 + 20\log D_0 - 20\log D_S - 6 - 10\log NOM$ ... (Eq. 6)

Automatic microphone mixers operate to keep the potential acoustic gain expressed by Eq. 6 constant at all times. In other words, automatic microphone mixers automatically insert attenuation that is equal to 10logNOM so that the total system gain when multiple mics are open and the potential acoustic gain when one mic is open are equal. Therefore, if one open mic does not cause the system to feedback, multiple open mics will not cause feedback either.

There are two types of automatic microphone mixers: gating and gain sharing.

4. Gating Automatic Microphone Mixers

An important element of automatic microphone mixers is the algorithm used to detect which mic is receiving valid voice input. Gating automatic microphone mixers use individual noise gates at each mic input for this purpose.

This type of automixer defines mics that exceed the gate threshold as open mics.

When a mic is simply turned on by a gate, a pop noise may occur because of the interruption of the low frequencies. In addition, when the mic is turned off immediately after a person stops speaking, the reverberation is abruptly cut off. With a well designed automatic microphone mixer, the attack time is increased sufficiently to prevent pop noise from occurring, the hold time is set appropriately so that the gate remains open while the person is speaking, and the mic is faded out after the voice is completely silent.

A gated system does not operate well if the threshold level is fixed. For example, if the threshold level is set too low, the ambient noise will cause the noise gates to open resulting in multiple open mics. If the threshold is set too high, syllables will be dropped. The right threshold for speech when the room is quiet will be wrong when the audience applauds or music plays.

To solve this problem, some models have a feature that dynamically sets the noise gate threshold level in accordance with the ambient noise. Various methods have been developed to increase the accuracy of the threshold level control, such as an adaptive type that constantly compares continuous ambient noise (such as noise from an air conditioner) to the sound that includes both ambient noise and amplified sound.

For example, a separate mic is used to measure the ambient noise level. When the speaker's mic input exceeds the ambient noise level, the mic is opened.

Gating automatic microphone mixers have a characteristic feature called the Last Microphone Hold. With this feature, the last used mic is left open until another mic is opened. It prevents word endings from being cut off unnaturally and also prevents reverberation and ambient noise from being cut off. The quality of the ambient noise may fluctuate until another person starts speaking, but it sounds more natural than ambient noise being cut off completely whenever there is a pause.

In typical gating automatic microphone mixers, to maintain constant potential acoustic gain, the NOM attenuation is inserted in the output stage after the mixing stage, as shown in the figure below. As mentioned earlier, the NOM attenuation level is 3 dB when there are two open mics, 6 dB when there are four open mics, and 9 dB when there are eight open mics.

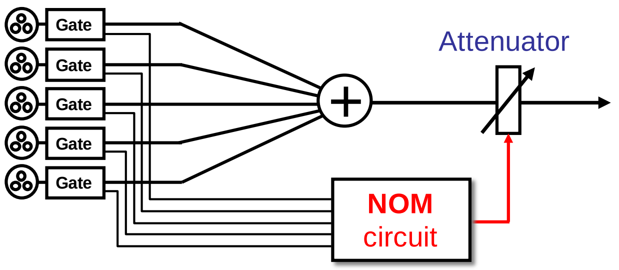

Figure 3

Some models have a feature that allows only one mic to be opened for a given sound source. This feature prevents not only gain errors but also overlapping mic input.

5. Gain Sharing Automatic Microphone Mixers

Gain sharing automatic microphone mixers use an algorithm developed by Dan Dugan. The algorithm compares each input with the sum of all inputs and adjusts the gain of each mic so that the potential acoustic gain of the sum of all mics is constantly equal to that of one mic. Because a constant gain is shared among the mics, this scheme is called "gain sharing." Because noise gates are not used, the beginning of sounds is never cut off. There is no need to configure complicated settings such as threshold levels, attack times, and hold times.

Signals input to a gain sharing automatic microphone mixer are split into a main audio path and a control side-chain. The main audio paths route the input signals to the outputs. The side-chain inputs are mixed to create a premix, which is used to compute the ratio of the level of that input to the level of the premix, the sum of all the inputs. That ratio becomes the gain of the channel. (Dan Dugan refers to this gain as "automix gain") Implicitly, this computation results in the total gain staying constant at all times. A conceptual diagram is shown in Figure 4.

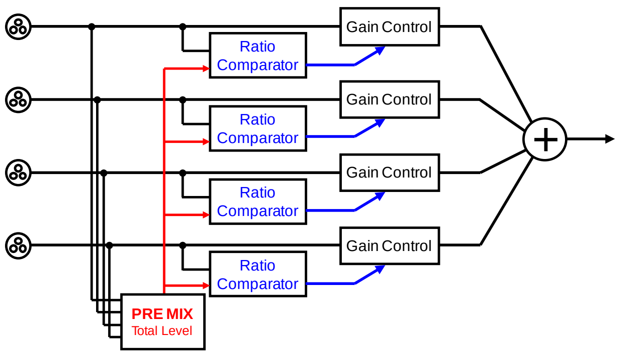

Figure 4

The automix gain of each input can be calculated using an amazingly simple formula:

Channel automix gain

$$= \frac{\text{level of channel input}}{\text{level of sum of all channel inputs}}$$

.................................(Eq. 7a)

If the levels are expressed in decibels, division is simply subtraction of logs; thus:

Channel automix gain (dB) = level of channel input – level of sum of all channel inputs ....(Eq. 7b)

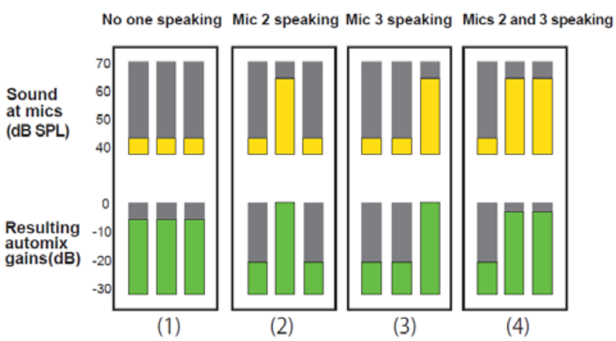

Figure 5

The effect of this algorithm with multiple mic channels will be explained using Figure 5. Unlike gating automixers, which turn mics on and off, gain sharing automixers seamlessly allocate appropriate gain to each input. This operation is similar to a situation in which a mixing engineer is accurately and swiftly controlling multiple input faders.

(1) represents a situation in which no one is speaking. The automix gain of each mic is the same, and the sum of automix gain of all mic inputs is equal to 0 dB, the gain of one mic.

(2) represents a situation in which one person is speaking into mic 2. The automix gain of the mic receiving the voice input automatically turns up to full, and the gains of other mics are automatically turned down.

(3) represents a situation in which another person starts speaking into mic 3. The gain automatically shifts to the mic receiving the voice input, and the gains of other mics are turned down. Since the gain is instantly and seamlessly reallocated, the beginning of the dialog is not cut off.

(4) represents a situation in which two people are speaking simultaneously. The gain is shared between the two mics receiving the voice input, and the unused mics are turned down. Because the total automix gain is the same as when there is only one open mic, feedback does not occur even when two mics are open simultaneously.

Here we will check situation (2) of Figure 5 by substituting values in Eq. 7b.

In this example, the ambient noise level (= the input level of mics 1 and 3) is 45 dBSPL, and the input level of mic 2 is 65 dBSPL. First, the sum of these three signal levels is $10\log(10^{45/10} + 10^{65/10} + 10^{45/10}) = 65$ dBSPL. We will substitute these values in Eq. 7b.

Automix gain for mics 1 and 3

$$= 45 – 65 = -20 \text{ dB}$$

Automix gain for mic 2

$$= 65 – 65 = 0 \text{ dB}$$

The sum of the signal levels after automixing all mics is shown below. It is equal to the signal level of one mic (mic 2).

Sum of the signal levels after automixing all mics

$$= 10\log(10^{(45-20)/10} + 10^{(65-0)/10} + 10^{(45-20)/10}) = 65 \text{ dBSPL}$$

Likewise, we will check situation (4) of Figure 5. The ambient noise level and mic input level are assumed to be the same as the previous case.

Automix gain for mic 1

$$= 45 – 68 = -23 \text{ dB}$$

Automix gain for mics 2 and 3

$$= 65 – 68 = -3 \text{ dB}$$

Sum of the signal levels after automixing all mics

$$= 10\log(10^{(45-23)/10} + 10^{(65-3)/10} + 10^{(65-3)/10}) = 65 \text{ dBSPL}$$

From this result, we can see that an attenuation equal to the NOM attenuation (NOM attenuation = 10log1 = 0 dB in situation (2) and 10log2 = 3 dB in situation (4)) is inserted in the input stage of the mic receiving the voice input.

A properly-implemented gain-sharing automixer treats correlated inputs in a special way. As an example, we will check the case when there are two input signals at 65 dBSPL. We substitute this value in Eq. 7. If there is no correlation between the two signals, the sum is equal to the original signal level + 3 dB. If there is correlation, the sum is equal to the original signal level + 6 dB.

- If there is no correlation

The automix gain is 65 – 68 = -3 dB. Each input signal will be attenuated by 3 dB. The sum of two mutually uncorrelated signals that have been adjusted to 65 – 3 = 62 dBSPL is 65 dBSPL, and this is the same as the signal level of one mic.

- If there is correlation

The automix gain is 65 – 71 = -6 dB. Each input signal will be attenuated by 6 dB. The sum of two mutually correlated signals that have been adjusted to 65 – 6 = 59 dBSPL is 65 dBSPL.

Thus gain sharing automixing evens out the level increase that normally would occur when the same signal hits two mics in close proximity to each other. This effect has a favorable application in the typical x-y gooseneck mic arrangement commonly seen on speakers' lecterns. Gain sharing automixing will even out the pickup pattern as the speaker moves left and right.

In a gating automixer, mics that are not receiving valid input signals are automatically turned down, typically 15 dB. In a gain-sharing automixer, the automix gain is reduced by a precise amount, computed to maintain the total gain in the system constant.

As mentioned earlier, with gating automixers, to prevent the awkwardness that results when reverberation after a dialog is cut off, the last mic hold feature is used to actively amplify the ambient noise even after the dialog is finished.

Gain sharing automixers do not require the last mic hold feature. This is because, regardless of the number of active mics, the signal level of the ambient noise is adjusted to match the ambient noise level picked up by one mic. In situation (1) of Figure 5, if we calculate the sum of the signal levels after automixing all mics, we can see that it is equal to the signal level of one mic as shown below.

The ambient noise level is assumed to be 45 dBSPL (the input level to each mic).

Automix gain for mics 1, 2, and 3

$$= 45 - 50 = -5 \text{ dB}$$

Sum of the signal levels after automixing all mics

$$= 10\log(10^{(45-5)/10} + 10^{(45-5)/10} + 10^{(45-5)/10}) = 45 \text{ dB}$$

In other words, the gain sharing algorithm inherently has a feature equivalent to the last mic hold feature of gating automixers.

In recording and broadcasting, natural ambience is essential. Automatic microphone mixers automatically retain the pickup of natural ambience. Therefore, with gain sharing automixers, mics can be left potted up, ready for use instantly.

The way in which overlapping mic sounds are handled on gain sharing automixers is also different from that on gating automixers.

In Figure 1, if the signal input level of the mic 1 m away from the speaker is 65 dBSPL, the input level of the mic 2 m away will be 65 – 6 = 59 dBSPL due to the attenuation over distance rule according to the inverse square law. The sum of these two signal levels is $10\log(10^{65/10} + 10^{59/10}) = 66$ dBSPL.

We substitute these values in Eq. 7 to determine the automix gains for the two mics.

Automix gain for the mic 1 m away

$$= 65 - 66 = -1 \text{ dB}$$

Automix gain for the mic 2 m away

$$= 59 - 66 = -7\text{dB}$$

The difference was originally 6 dB, but the difference after passing through the automatic microphone mixer is 12 dB. From this, we can see that the mic separation has increased. If the original difference is 3 dB, the difference after passing through the automatic microphone mixer will be 6 dB. If the original difference is 9 dB, the difference after passing through the automatic microphone mixer will be 18 dB. We obtain twice the separation.

This minimizes sound quality degradation caused by comb filters and enables clear sound amplification not only in conference situations but also in lectures that have two mics on stage.

6. Dan Dugan Sound Design

Dan Dugan Sound Design is a company headed by Dan Dugan. The company is famous for its gain sharing automatic microphone mixers. The automatic microphone mixers are well known in broadcasting stations and conferences especially in North America. Originally the company developed and manufactured analog computing automatic microphone mixers, but in the recent years, they are developing DSP mixing controllers that are inserted into the input signal paths of mixing consoles.

After six years of research, Dan Dugan developed the gain sharing algorithm in 1971, which later received a patent. This algorithm was used in Altec Lansing automatic microphone mixers. The engineering design has gone through many improvements since then, and today Dan Dugan Sound Design's automatic microphone mixers have become a standard in programs that use multiple microphones in North America.

7. Dugan-MY16

The Dugan-MY16 card is a product made by Dan Dugan Sound Design for Yamaha's digital mixing consoles. The card incorporates the company's latest gain sharing algorithm.

The card can be installed in a Mini-YGDAI card slot of Yamaha's digital mixing consoles. It is recommended that the card be inserted post fader in input channels. One card supports automatic mixing of 16 channels (at 48 kHz sampling). The maximum number of cards that can be used simultaneously is eight (128 channels at 48 kHz sampling).

Each input can be assigned to one of three groups (a, b, and c), and each group functions as an individual automatic microphone mixer. The card has features such as weighting, mute, bypass, and override. It can be controlled and monitored from a PC, Mac, or iPad.

8. Conclusion

As explained in this document, automatic microphone mixers are different from automatic gain controllers, compressors, or simple noise gates. Gain sharing automatic microphone mixers allow mixing engineers to actively adjust the level balance between mics on the mixing console while not having to worry about cueing mics in and out. They help engineers to mix productively with confidence while maintaining safe operation and high quality sound.

※ We do not recommend that you use gain sharing automatic microphone mixers in music sound reinforcement applications. Music mixing requires artistic balance. Gain sharing automatic microphone mixers are designed to serve a different purpose (as described in this document). They do not necessarily achieve musical balance automatically.

Bibliography

(2. Acoustic Gain, 3. Effects of Multiple Open Mics)

*1 Sound System Engineering, Don Davis & Carolyn Davis

Yamaha Corporation

Pro Audio Division

10-1, Nakazawa-cho, Naka-ku, Hamamatsu, Shizuoka, Japan

http://www.yamahaproaudio.com/