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AUTOMATIC MICROPHONE MIXERS: SOLVING THE AUDIO PROBLEMS CAUSED BY MULTIPLE OPEN MICROPHONES

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1. INTRODUCTION

High quality audio becomes progressively more difficult to achieve as the number of open microphones increases. All audio systems face the same problems whenever multiple open microphones are employed in the same acoustic environment. These problems are:

- Build-up of background noise and reverberation
- Reduced gain before feedback
- Comb filtering

All of the problems above can plague any audio system where multiple microphones are used, e.g., boardrooms, city council chambers, conference centers, houses of worship, broadcast studios, teleconferencing rooms. Since audio quality rapidly deteriorates as the number of open microphones increases, the solution is to keep the minimum number of microphones open that can appropriately handle the audio. This can be accomplished by having each microphone user activate and deactivate his microphone via a local switch; or by having an audio engineer activate and deactivate microphones at a central control console; or by employing an automatic microphone mixer. Automatic microphone mixers, introduced in the mid-1970's, are designed to keep unused microphones attenuated, and to activate any microphone spoken into within milliseconds.

2. BUILD-UP OF BACKGROUND NOISE AND REVERBERATION

The first problem of multiple open microphones is the build-up of background noise and reverberation. This build-up adversely affects the quality of recordings or broadcasts originating from the audio system. Consider the case of a roundtable discussion with eight members and eight microphones. If all eight microphones are open, when only one microphone is needed, the audio output will contain the background noise and reverberation of all eight microphones. More precisely, the audio output from eight open omnidirectional microphones would contain 9 dB more background noise and reverberation than a single omnidirectional microphone. To the human ear, the background noise and reverberation would be perceived as almost twice as loud when all eight microphones were open. This build-up of background noise and reverberation greatly deteriorates speech clarity and intelligibility.

To minimize background noise and reverberation build-up, an automatic microphone mixer attempts to activate only the microphone(s) being addressed <u>and</u> employs a "NOMA" circuit. "NOMA" is an acronym for <u>N</u>umber of <u>O</u>pen <u>M</u>icrophones <u>A</u>ttenuation. NOMA systematically <u>decreases</u> the audio output whenever the number of open microphones increases. Without NOMA, the audio system would produce objectionable noise modulation ("pumping and breathing") as background noise and reverberation increase and decrease with the number of open microphones. With a well-designed automatic microphone mixer, background noise and reverberation remain constant no matter how many or few microphones are activated.

The accepted attenuation rate of NOMA is 3 dB for every doubling of the number of open microphones. Stated another way:

Level of Attenuation = 10 log (Number of Open Microphones)

An unwanted side effect of NOMA is that the level of a talker will be lowered if an additional microphone is accidentally activated by a random sound such as a cough. But the talker's original level will be restored once the additional microphone is deactivated.

The following table lists a variety of miking configurations for a round table discussion with eight participants. Note that eight cardioid microphones into an automatic mixer will exhibit almost 14 dB less ambient noise pickup than eight omnidirectional microphones into a non-automatic mixer.

Table 1: ROUND TABLE DISCUSSION
TECHNIQUES FOR MIKING EIGHT PARTICIPANTS

NUMBER OF MICROPHONES	MICROPHONE TYPE	DISTANCE TO TALKER	AUTOMATIC MIXING	AMBIENT NOISE PICKUP
8	omni lavalier	12"	no	0 dB
4	cardioid table stand	18"	no	-3.0 dB
1	omni surface	30"	no	-4.1 dB
8	cardioid table stand	12"	no	-4.8 dB
4	supercardioid surface	18"	no	-5.6 dB
4	cardioid table stand	18"	yes	-9.0 dB
8	omni lavalier	12"	yes	-9.0 dB
8	cardioid table stand	12"	yes	-13.8 dB

courtesy of Bruce Bartlett / Crown International

3. REDUCED GAIN BEFORE FEEDBACK

The second problem of multiple open microphones is reduced gain before feedback. Acoustic feedback or "howling" can often be a problem anytime a sound reinforcement (PA) system is used. To avoid feedback, PA systems are operated below the point where the system becomes unstable and starts to feed back. However, this feedback safety margin is reduced each time another microphone is opened. Have one too many open microphones and the result is feedback.

As in the previous section, the automatic microphone mixer solution is to keep unused microphones attenuated and utilize a NOMA circuit. As more microphones are activated, the overall gain will remain constant thanks to the NOMA circuit. Using an automatic microphone mixer, if the audio system does not feedback when any one microphone is open, the system should remain feedback-free even if all the microphones are open.

The formula for the loss of available gain before feedback is:

$$\text{Loss of Gain before Feedback} = 10 \log (\text{Number of Open Microphones})$$

An automatic microphone mixer essentially keeps the number of open microphones electronically equal to one via the NOMA circuit. Therefore, the above equation equals zero as the log of one is defined as zero; in practice, there is no loss of gain before feedback when a well-designed automatic microphone mixer is employed in a PA system. Gain before feedback will then be determined primarily by the variables in the Potential Acoustic Gain equation and its many variations.

Table 2: LOSS OF GAIN BEFORE FEEDBACK

NUMBER OF OPEN MICROPHONES	LOSS OF GAIN
1	0 dB
2	- 3.0 dB
3	- 4.8 dB
4	- 6.0 dB
5	- 7.0 dB
6	- 7.8 dB
7	- 8.5 dB
8	- 9.0 dB

4. COMB FILTERING

Comb filtering occurs when open microphones at different distances from a sound source are mixed together. See Figure 1. Since sound travels at a finite speed, the sound waves from the source arrive at the microphones at different times. As a result, the outputs of the microphones are not in phase with each other. When combined in a mixer, these out-of-phase microphone signals produce a combined frequency response very different from the frequency response of any single microphone.

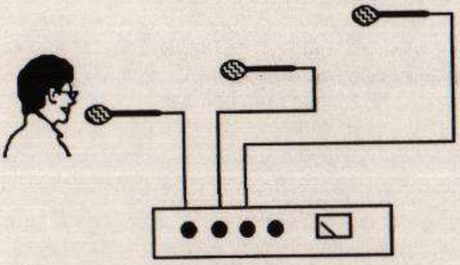

Figure 1

Comb filtering occurs when open microphones at different distances are mixed together.

To illustrate how comb filtering deteriorates an audio signal, a series of frequency response measurements were made using three microphones of the same model. All measurements were made in an anechoic chamber.

The frequency response curve of a Shure model SM80 omnidirectional condenser microphone is shown in Chart 1. The SM80 was placed two feet away from the source loudspeaker. Note the flat, smooth response curve of the single SM80 microphone.

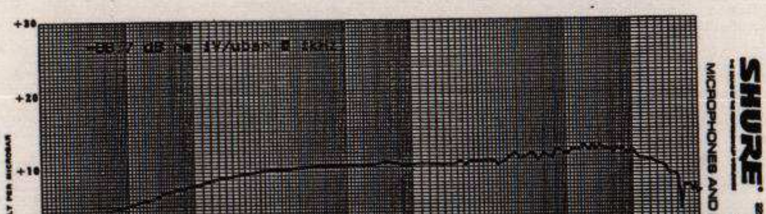

Chart 1

Omnidirectional condenser microphone two feet from sound source

Chart 2 is the combined frequency response of two SM80 microphones. One microphone was two feet from the loudspeaker and the other was four feet. The gain settings on the mixer were set the same for both microphones. Comb filtering is easily seen in this response curve.

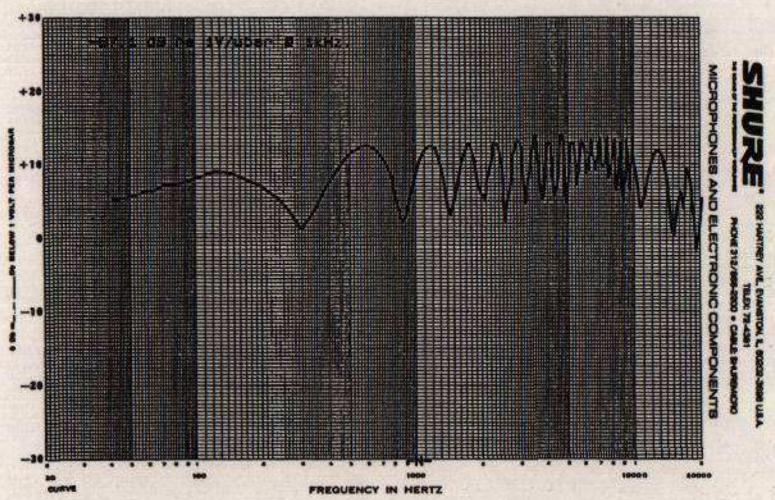

Chart 2
Two omnidirectional condenser microphones; two feet and four feet from sound source

Chart 3 is the combined frequency response of three SM80 microphones. The microphones were placed two feet, four feet and six feet from the loudspeaker. As before, the gain settings on the mixer were set the same for all three microphones.

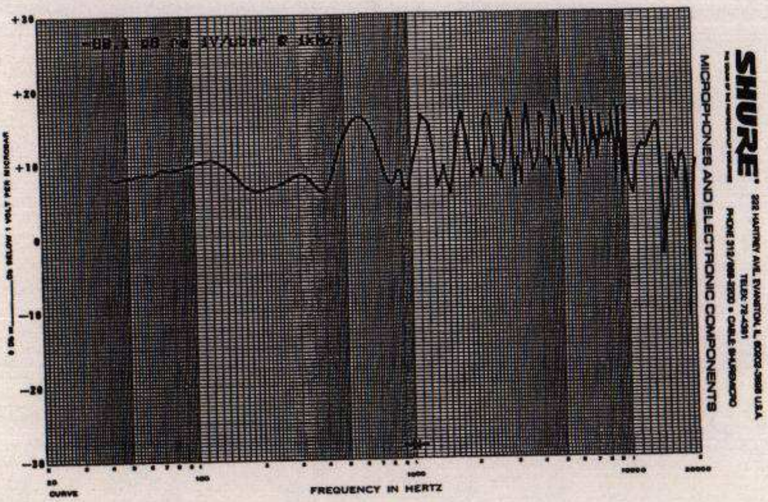

Chart 3
Three omnidirectional condenser microphones; two feet, four feet, and six feet from sound source

The aural result of comb filtering is an audio signal that sounds hollow, diffuse, and "phasey." Automatic microphone mixers reduce comb filtering by keeping unused microphones attenuated. It is not necessary to keep unused microphones completely off to gain the aural advantages of automatic mixing. On most automatic microphone mixers, the amount of attenuation applied to a microphone when it is not activated is adjustable. Typical adjustments range from 8 dB of attenuation to completely off. An attenuation level of 15 dB has been found to be satisfactory for most situations. Keeping a microphone attenuated 10 to 15 dB instead of completely off makes the activation of that microphone sound smoother. Well-designed automatic microphone mixers typically take less than five milliseconds to raise a microphone from the attenuated "off" state to the in-use unattenuated "on" state.

5. CURRENT ACTIVATION TECHNIQUES OF AUTOMATIC MIXERS

At its core, an automatic mixer must quickly differentiate when a person is speaking into a microphone and when a person is not speaking into a microphone. When the mixer decides that a person is speaking, it must rapidly raise that input channel from an attenuated state ("gated off") to an unattenuated state ("gated on"). That is the essence of what an automatic mixer must do well; how it makes that decision is the subject of multiple patents. An in-depth discussion of the various techniques would require a much lengthier paper, so the following descriptions are accordingly brief.

Fixed Threshold: This technique requires that a signal level threshold be set for each mixer channel, and that this threshold be exceeded by the talker's signal level for the input channel to activate. Basically, this type of technique is a variation of a noise gate. Two fundamental problems plague a fixed threshold automatic mixer. one - how to initially set thresholds when talkers are not present; two - a fixed threshold technique is not useful if the ambient noise conditions vary, such as when an audience applauds.

Adaptive Threshold using Ambiance Microphones: This technique uses a single microphone, centrally located, to raise or lower the overall threshold based on instantaneous ambient noise conditions. A variation of this technique is to employ a special dual element directional microphone, one element facing the talker (signal) and the other facing away (ambient), to dynamically set the threshold based on local ambient noise conditions at the microphone location. This dual element technique also provides activation that is directionally sensitive, i.e., only sounds within a certain angle, typically 120°, will cause the automatic mixer input to activate.

Adaptive Threshold using the Talkers' Microphones: This technique employs an average of all the input levels and then activates any input(s) which exceeds the average by a predetermined amount. Typically, a minimum activation threshold level is added to the average of all the input levels. A variation of this technique is to compare an input level to an average that does not include that input signal. In this variation, the threshold for any input is determined by all the other inputs.

Noise Adaptive Threshold: This technique uses an activation threshold unique to each input that is determined from each individual input signal, not from an average of all input signals. Essentially, this unique input threshold is determined by using an inverse peak detector, i.e., a circuit that detects the minimum levels of the input signal instead of the maximum levels. When the input signal level detector exceeds the minimum threshold level by a preset amount and the input level is transient in nature, like speech, then the mixer input is activated. When multiple inputs detect the same talker, a comparator circuit activates only the input which detects the strongest signal from that talker. In most situations, this technique is not appropriate for music because long sustained notes might be detected as constant ambient noise and cause the mixer input to deactivate.

Level Proportional Gain: This technique varies the available gain of the mixer based on the comparison of all input levels. If all input levels are equal (no one is talking), the gain of the mixer is shared equally among all inputs. When one input level exceeds all others (one person is talking), that input receives full gain and all other inputs are reduced in gain, yet the total gain is equal to when no one is talking. When the levels of two inputs exceed all others (two people are talking), these two inputs each receive full gain minus 3 dB and the other inputs are reduced in gain proportionally. To summarize this technique, the available gain of the mixer is shared among all channels based on the levels detected at the mixer inputs.

Scanning Threshold: This technique compares each input's peak signal level to a dynamic threshold that constantly moves downward in level. A complete cycle of the threshold from maximum to minimum is less than 10 milliseconds. When an input's peak level meets or exceeds this moving threshold, the mixer activates that input channel for a short time period, typically 200 milliseconds. The dynamic threshold is then reset to it's maximum level and immediately begins to move downward again. If the threshold encounters the same input's peak level, the activation time of that input is extended and it remains active. If another input's peak level is encountered, that input is activated. More than one input can be active at any instant.

Perhaps the obvious question is which of these activation techniques is the best. It depends on whom you ask. All of the techniques are employed in current automatic mixer designs and all have their proponents. Until all manufacturers of automatic mixers employ the same technique, the question of which is best will remain a subject open to vigorous and enthusiastic debate among audio system designers.

6. SUMMARY

- Keeping the number of open microphones to a minimum always improves overall audio quality.

- The primary function of an automatic microphone mixer is to keep unused microphones attenuated (turned off) and to imperceptibly activate microphones when needed.

- Build-up of background and reverberant noise, reduced gain before feedback, and comb filtering can be all controlled by using a well-designed automatic microphone mixer.

- There is a wide variety of circuit techniques employed in commercial automatic mixers to determine when an input should be activated.

7. ADDITIONAL READING / REFERENCES

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