Dynamic Microphone Overload

In a recently published article the following statement is made:

"... the transistors are not overloaded. This is not a very serious problem when dynamic microphones are used, because their output is normally about −60 dBm and does not rise higher than about −35 dBm."

This statement points out a great fallacy in some users' understanding of the dynamic microphone. How many times have I heard a mixer, when listening to a playback, say, "listen to that dynamic microphone overload."

A properly designed dynamic microphone will not overload under any reasonable level. But, just what constitutes a reasonable level?

During the design and construction of the dynamic element, it is desirable to permit the maximum amount of movement possible without causing a loss of output level, or adding excessive mass to the coil or diaphragm. If the limit is set, such that the diaphragm will not touch or bottom until a certain sound level or pressure is exceeded, then any sound pressure up to this limit will not cause the microphone to overload and produce distortion of the waveform.

How can this be proved? And is it reasonable to expect a sufficiently high limit of sound pressure before bottoming in the microphone, so that it will not overload at our "reasonable level"?

I do not know exactly what procedures are used by other manufacturers, but at E-V each microphone design is checked with a 60 Hz sine wave in a closed cavity, at a sound pressure of 130 dB. It is mandatory that the microphone reproduce a clean sine wave at this level. Additional checks are made at higher levels, and at different frequencies to prove out the design under consideration. In addition, each professional microphone, as part of its manufacturing process is subjected to the same 60 Hz – 130 dB S.P.L. test, with the result being observed on an oscilloscope. The waveform must be clean or the particular unit microphone is rejected and scrapped.

A properly designed dynamic microphone thus will not overload under any reasonable level (130 dB S.P.L.).

What does this mean to the mixer in his use of the microphone? It means that for any musical pickup the microphone will perform with a clean pickup of the instrument or voice. The distortion we may experience is now due to the high (but clean) output of the microphone overloading the *input stage of the first amplifier*.

How is a microphone, rated at −55 dBm, going to overload a preamplifier with a maximum rated input of −22 dBm? (I am using the RCA BA-31 as an example.) First – the microphone is rated by applying a signal level, that being equal to 94 dB S.P.L. Second – the input rating of the amplifier is the absolute maximum input before overload is encountered. Unlike tube amplifiers that had a few dB leeway, and gave a small warning in slightly increased distortion, the typical transistor unit will reproduce a clean signal when fed with a sine wave at −22 dBm, and a clipped, distorted signal when hit with −20 dBm. Again, unlike the tube amplifier which had a soft shoulder the transistor amplifier will evidence sudden, severe distortion when subjected to overload. There is no safety margin and so one must be created by the user.

In a typical session of today's sound, the instruments are loud, and the vocals often screaming. For maximum separation and reduction of acoustical phase distortion, the mixer is forced to place the microphones close, often within a few inches. With this type of pickup, the impact to the diaphragm is often on the order of 120 to 125 dB. S.P.L. average level, and hard hits, or 'beats' can push this to 130 dB S.P.L.

With this kind of level, let's re-examine the output of the microphone. The typical −55 dBm rated microphone can, in these cases, put out a signal of −19 dBm, ... 3 dB over the maximum rated input of the amplifier! Should the output of the microphone be merely

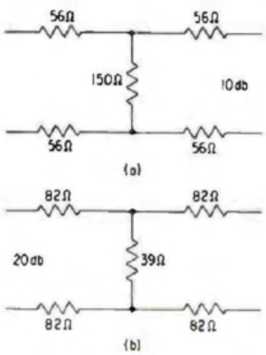

Fig. 2. Balanced H pads designed for 150-ohms impedance. (a) shows the values for a 10 dB loss; (b) shows 20 dB.

equal to the maximum input of the amplifier, what is going to keep it at this level, and what has happened to the musical transients? The solution is simple, reduce the input level to the amplifier. Now we have our safety margin.

Some consoles are providing input attenuation, by means of selectable pads, in line, before the first amplifier. If your console does not have such a device, a single pad (switchable to preserve signal-to-noise ratio when you do not need the reduction) can be purchased in fixed units of reduction, either for in-console or as plug-in devices in the microphone line. In FIGURE 2, values for a 10 dB and a 20 dB symmetrical, balanced H pad designed for impedances of 150 ohms are shown. To change impedance for your system, multiply the values given by the ratio of the desired impedance match divided by 150 (Z/150).

The selection of the proper amount of padding necessary is sometimes difficult, as it is desirable to maintain as high an input level to the amplifier as possible. This will preserve a good signal-to-noise ratio, and still maintain a margin for peaks and transients. I have found, for me, a good margin, by allowing 10 dB above the loudest planned input, and padding to the nearest convenient point.

When you are first getting to know a console and the various microphones, as well as a new microphone technique, it might be useful to utilize a wide range VU meter (−60 dBm to 0 dBm or above) to analyze the various input levels you are encountering.

In conclusion, if you have distortion due to overload (or suspect it) first check the input level to the amplifier, either by measuring or by heavily padding. If distortion still occurs, then suspect a damaged dynamic microphone.