GATES
ENGINEERING
REPORT

UNITS OF MEASUREMENT IN EQUIPMENT PERFORMANCE

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GATES

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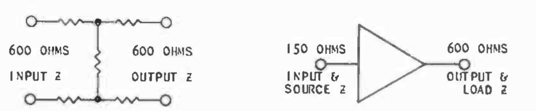
The Broadcasting Industry employs many technical terms in papers, reports, brochures, text books, conversation and other communications to accurately convey information about components, circuits, equipments and systems. Although the definition of most of these terms may be found in various publications, the exact interpretation and application may differ sufficiently in current usage to cause considerable confusion. Ironically, many of the most commonly used terms or units of measurement are those with the most interpretations. An explanation of some of the units of measurement in equipment performance may be helpful for those engaged in specifying new equipments and/or systems design.



TYPICAL MATCHING ATTENUATOR

VOLTAGE RATIO: 100:1
CURRENT RATIO: 100:1
POWER RATIO: 10000:1
ATTENUATOR LOSS: 40 db

TYPICAL MICROPHONE PREAMPLIFIER

VOLTAGE RATIO: 1:200
CURRENT RATIO: 1: 50
POWER RATIO: 1:10000
AMPLIFIER GAIN: 40 db

THE DECIBEL, ABBREVIATED db, IS A UNIT USED TO EXPRESS THE RATIO BETWEEN TWO AMOUNTS OF POWER (P_2 AND P_1) EXISTING AT TWO POINTS:

NUMBER OF db = $10 \log_{10} P_2/P_1$ FOR POWER RATIOS.

WHEN THE POINTS OF MEASUREMENT ARE OF IDENTICAL IMPEDANCE ONLY:

NUMBER OF db = $20 \log_{10} E_2/E_1$ FOR VOLTAGE RATIOS.

NUMBER OF db = $20 \log_{10} I_2/I_1$ FOR CURRENT RATIOS.

VOLTAGE, CURRENT AND POWER RATIOS

FIGURE 1

GAIN is considered to be the power gain of the equipment or system at 1 KHz, unless otherwise qualified. Two common ratings are the maximum gain, where all attenuators are set for minimum attenuation; and the normal operating gain, where all attenuators are set in their optimum operating position. With two or more attenuators in series in a system, the optimum operating range of each of them must be established and followed for the best results over the dynamic range of the system.

DECIBEL - Gain is normally expressed in db as the ratio of the input power to the output power of the equipment or system. A power ratio of 2 gives 3 db power gain, of 4 gives 6 db power gain, of 10 gives 10 db of power gain, etc. The impedance of the two points compared is relatively unimportant for a measurement of power gain, if the readings are converted to a common base such as RMS power, peak power, etc.

Since many power gain measurements are actually conversions from voltage measurements, considerable chance for error or misinterpretation exists. Voltage gain may be obtained from passive components such as step-up transformers - where the power gain would be less than unity, due to circuit losses. Conversely, power gain may be obtained from active circuits such as emitter-followers - where, due to inherent characteristics, the voltage gain would be less than unity. Thus, the power gain is directly related to voltage gain only when the impedances of the points of comparison are equal. Then, a voltage ratio of 2 will give a power gain of 6 db, of 4 will give a power gain of 12 db, of 10 will give a power gain of 20 db, etc.

The term "db voltage gain" is sometimes used to specify the voltage ratio of two points of unequal impedance. It would be more correct to use "the voltage equivalent of (blank) db gain" instead. Thus, "the voltage equivalent of 20 db (power) gain" would specify that there was a voltage ratio of 10 times between two points of presumably unequal impedance.

DBM - A basic tool of the Broadcast Industry is the term dbm, which has a specific base of reference of 0 dbm for one milliwatt across 600 ohms impedance. It can be applied only to recurrent or periodic waves of a sinusoidal nature, to make it a truly universal reference level. It can be measured on any accurate RMS voltmeter and/or a properly calibrated volume indicator (meter plus attenuator). Impedance transformations or conversions allow the use of dbm on impedances other than 600 ohms, as long as there is a direct power ratio between the points of measurement.

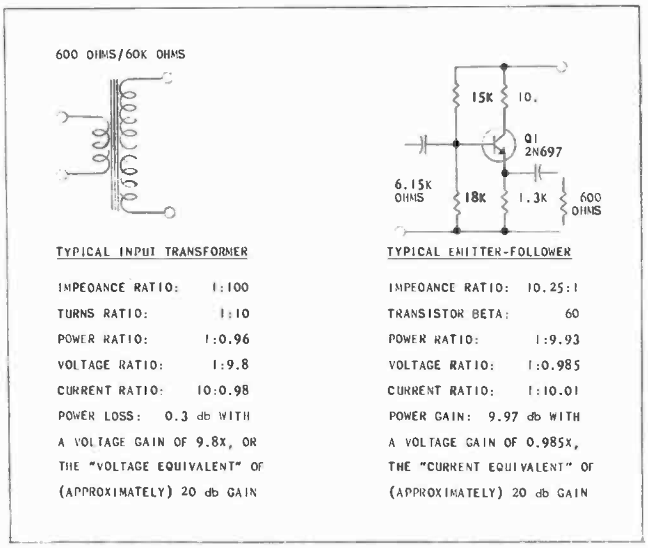


FIGURE 2

The term dbm is particularly useful in specifying the absolute level or capability of any point in a system - such as the threshold of noise, threshold of limiting, maximum input or output level, etc. Properly interpreted, it will give the experienced engineer an excellent indication of the potential behavior of a system or equipment under complex wave conditions over the extreme dynamic range of programming.

VU - Although the volume unit has the same specific base of reference as dbm - 0 vu for one milliwatt across 600 ohms impedance - this is the end of the relationship. While the term dbm can be applied only to recurrent or periodic sinusoidal waves, the term vu can be applied only to nonrecurrent or nonperiodic waves of a complex nature, such as found in speech and music. ASA C16.5-1954, paragraph 3.9 defines: "The reading is determined by the greatest deflections occurring in a period of about a minute for program waves, or a shorter period (e.g., 5 to 10 seconds) for message telephone speech waves, excluding not more than one or two occasional deflections of unusual amplitude."

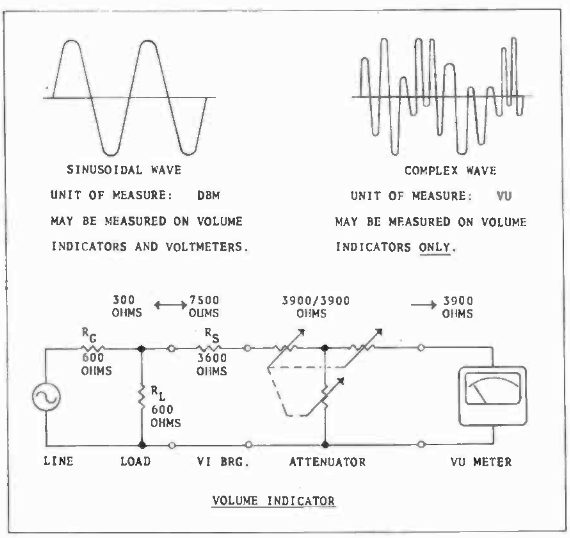


FIGURE 3

The ballistic characteristic or dynamic behavior of a volume indicator under prescribed conditions is clearly defined by the referenced ASA Standard. A volume indicator meeting these standards will give uniform readings for many types of programming. It can, however, give greatly different readings for two dissimilar complex waves with the same peak amplitudes. Thus, the level in vu may be determined more by the volume indicator characteristics than the peak amplitude of the signal being measured. This can be readily observed by comparing the volume indicator readings with an oscilloscope display with different types of programming.

HEAD ROOM - All amplifiers and components in audio systems should have a peak power rating considerably greater than the normal operating power level, in order to prevent serious distortion on complex programming peaks. The ratio of these complex wave peaks to the volume indicator reading is much more than the peak to RMS power ratio of a sinusoidal wave (which is 2:1 or 3 db). The peak factor of a complex wave is generally considered to range up to 10 db with much of the pre-recorded material - where pre-processing has reduced excessive peaks. This peak factor can range up to 20 db or more with certain microphone techniques on some "live" material.

Preamplifiers employed before the first fader in a system must be able to handle this excessive head room, plus exceedingly high levels in dynamic range. Thus, microphone preamplifiers with a maximum input level capability of –17 dbm and output capability of +25 dbm can be fully utilized in today's broadcasting and recording.

There is some preference among systems engineers to label systems block diagrams with the normal complex wave levels in vu - expecting the equipment manufacturer to know the anticipated peak factor, and to provide sufficient head room. It may be better to specify the normal levels and maximum levels in dbm (since most testing is performed with sinusoidal waves), to specify the amount of head room desired.

The standard telephone line feed of +8 vu is generally considered to require a line amplifier with an output capability of +24 dbm; which is reduced to +18 dbm by the 6 db line isolation pad. Although there is no actual relation between +8 vu and +18 dbm (since vu applies to complex and dbm to sinusoidal waves only), the head room is considered to be 10 db. For systems anticipating an appreciable amount of "live" programming, it would be better to consider line amplifiers with a higher output capability.

Some systems are using line amplifiers with +32 dbm output capability. This is reduced to +26 dbm by the 6 db isolation pad, for a program peak factor or head room of 18 db. Program peaks of greater amplitude are nearly always narrow "spikes" that can be clipped off in the amplifiers without any discernible degradation - since they will cause no base line shift, etc., in a properly designed amplifier.

INPUT IMPEDANCE is a term that is frequently confused with source impedance, but the two are distinctly different. Input impedance refers to the loading that the input of an amplifier, or a passive circuit such as a filter, will place across a signal source. Many amplifiers, such as microphone preamplifiers, have an input impedance of 10 times or more than their rated source impedance.

SOURCE IMPEDANCE refers to the impedance of the signal source at the point under consideration. When feeding an amplifier a test signal from an oscillator/attenuator panel - the output of the oscillator/attenuator unit would present a certain source impedance to the input of the amplifier. The output of a console mixing bus will present a constant source impedance to the input of the booster or line amplifier.

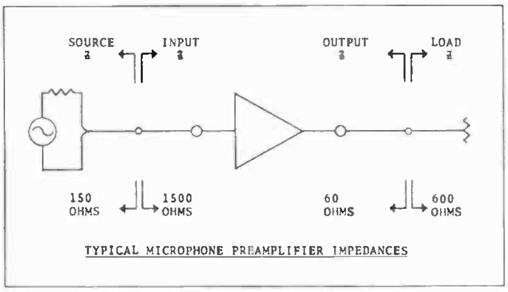


FIGURE 4

OUTPUT IMPEDANCE is often confused with the load impedance of a device, especially with an amplifier. A more descriptive term is "reflected output impedance" - the output impedance is determined by the internal impedance of the output section of the device being described. Amplifiers with negative voltage feedback from the output stages normally have an output impedance that is a fraction of the rated load impedance - generally one tenth or less.

The output impedance of a device is an important consideration in systems design. Attenuators and most passive networks have an output impedance that approximates their load impedance. They generally exhibit a difference of twice voltage or "the voltage equivalent of 6 db" on the output between the unloaded and loaded conditions. They are sensitive to bridging loads. For example, when the 7500 ohm bridging impedance of a standard volume indicator is switched across a circuit with 600 ohms output impedance, terminated with a 600 ohm load, the loading effect is approximately 0.4 db. However, if the volume indicator is switched across the output of an amplifier with an output impedance that is 1/10th the rated load impedance of 600 ohms, the loading effect is almost imperceptible.

LOAD IMPEDANCE is the proper term to use to specify the terminating impedance of an equipment whether it is an active or passive device. The load impedance of an amplifier is the impedance that should be placed across the output terminals to obtain its rated characteristics. In systems employing the input of amplifiers, or other devices with an input impedance that is higher than the source impedance (as loads for preceding circuitry), it is common practice to parallel the amplifier input with a resistor that will give a parallel combination that equals the desired impedance. Some components do not require termination or should not be terminated with their rated impedance. Thus, the input of most microphone preamplifiers present a bridging impedance to the output of the microphone - taking advantage of the double voltage output of the unloaded microphone, for approximately 6 db better signal-to-noise ratio.

OUTPUT TO INPUT ISOLATION is an important consideration where the output of an amplifier feeds a combining pad or network where different signals are mixed. In order to keep the signal pure on the input of the amplifier, it is necessary to have a high ratio of output to input isolation. Transistor amplifiers generally have lower isolation than tube amplifiers, especially those with negative feedback to the base of the first stage.

CONSTANT INPUT VERSUS CONSTANT OUTPUT LEVEL - This has been the subject of many an argument between customer inspectors and suppliers, etc., and an area that is seldom defined. Should the input level be held constant at the various frequencies under test, or should the output level be held constant? Of course, on many "hi-fi amplifiers" this is no problem, as they are rated "flat" from 3 Hz to 150 KHz or higher.

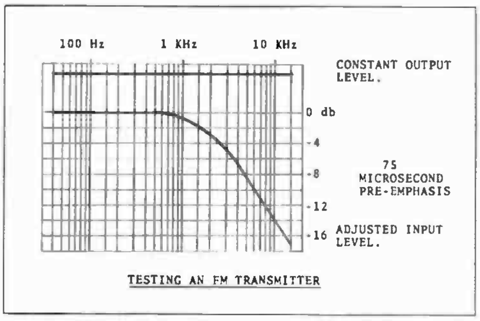


FIGURE 5

Unless otherwise qualified, it is good engineering practice to test harmonic distortion with a constant output level - in order to determine if the maximum output can be obtained at all signal frequencies. This is especially true with units, such as FM Transmitters, that have appreciable preemphasis networks incorporated in them.

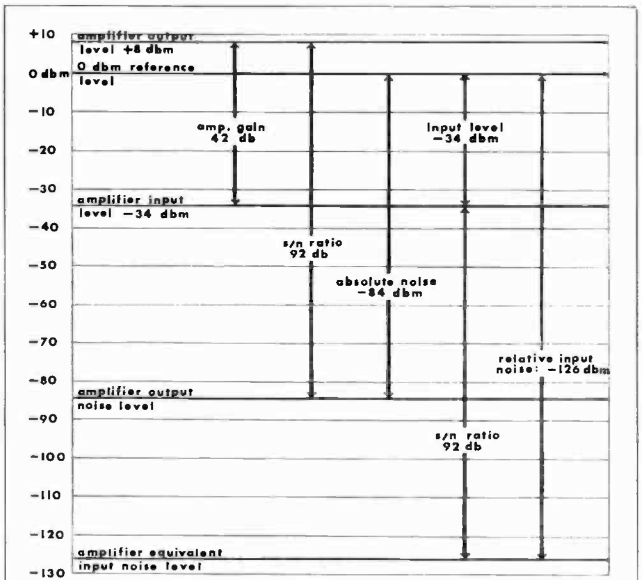
If the unit under test is kept below the overload point, there is little difference between using constant input or constant output level in testing frequency response. The decision is generally subject to the flexibility of the source and load measuring equipment. With some it is easier to read input level adjustments. With others it is the opposite. A rather common practice with audio equipment is to test the frequency response with a constant input at the normal operating level; and to test the harmonic distortion with a constant output level at the maximum capability.

HARMONIC DISTORTION is too often shortened to "distortion" although this is certainly not the only kind of distortion encountered in broadcasting. When the reference does not qualify the type of distortion, it is generally considered to be harmonic distortion. It occurs because of circuit non-linearities that create harmonically related signals in the output which were not present in the input of the equipment or system. Single tone testing with the fundamental tone nulled out allows direct reading of the sum of the harmonics on distortion tests.

IM DISTORTION is an accepted abbreviation for intermodulation distortion, created by mixing of two or more signals due to circuit non-linearities. The most pronounced products of IM distortion are the sum and difference frequencies of the fundamental frequencies. Two tone testing with a 4:1 ratio of tone amplitudes yields a commonly accepted measurement.

FREQUENCY DISTORTION is a term sometimes used to signify that the frequency response is other than "flat", or that it fails to faithfully follow some prescribed pre-emphasis or de-emphasis. Where it is used in lieu of the term frequency response; the upper and lower frequency limits, as well as the maximum deviation, should be specified.

SIGNAL PLUS NOISE TO NOISE RATIO, abbreviated S+N/N, is the noise measurement most frequently taken on an equipment or system, although it is often converted to one of the other noise measurement terms. In broadcast equipment, where the ratio of S+N/N is generally higher than 40 db, just the term signal-to-noise ratio is commonly used and is abbreviated S/N. It is normally measured down from a referenced dbm output level.



TYPICAL PREAMPLIFIER NOISE RELATIONSHIP

FIGURE 6

ABSOLUTE NOISE is the noise level in an equipment that is referred to 0 dbm. With an amplifier having 42 db gain (measured with an input level of -34 dbm and an output level of +8 dbm) the noise is found to be 92 db below the +8 dbm output reference. The same amplifier would yield a noise measurement of 84 db below a -42 dbm input and 0 dbm output level - which is the absolute noise level. Or, the absolute noise can be derived by subtracting the output above 0 dbm (8 db) from the 92 db measured below the +8 dbm output reference for a figure of -84 dbm.

RELATIVE INPUT NOISE, also referred to as equivalent input noise, is conveniently found by adding the input level of the equipment under test to the S/N measured on the output. Thus, the amplifier with -34 dbm input level and a signal-to-noise of 92 db can be converted by adding the two figures to obtain a relative input noise figure of -126 dbm. This term is particularly useful in a complex system to calculate the S/N at each critical point (generally at the amplifier input terminals). As long as the rated relative input noise of the amplifier allows a S/N of 8 to 10 db better than the S/N of the incoming signal, no noise degradation results in most systems.

WIDE BAND NOISE VERSUS WEIGHTED NOISE - Wide band noise, usually given as "noise", is normally considered to be that band between approximately 5 Hz and 50 KHz which is essentially flat in a modern noise analyzer unit. Both frequency limits greatly exceed the accepted limits of human hearing - which many rate from 30 Hz to 15 KHz maximum. Studies of the sensitivity of the human ear over the past several decades agree that the upper and lower regions of the audible spectrum are considerably attenuated, compared to the 2500 Hz region.. Thus, many noise measurements are more meaningful with a weighting curve that will partially compensate for the non-linearity of sensitivity versus frequency of the human ear. In some instances this curve may be derived from a high-pass and low-pass filter to restrict the limits to a more narrow band. Or, it may be obtained by partially attenuating certain sections of the frequency band compared to those considered more sensitive.

CROSSTALK is the result of signal (s) being fed into adjacent channels by capacitive, inductive and/or resistive coupling. Just as any plant in the wrong spot is considered to be a weed - crosstalk may be considered a form of noise. Since crosstalk is generally a very low level signal - and the human ear is very much more sensitive to mid-range frequencies than those in the upper or lower portion of the audio spectrum - mid-range crosstalk is the critical area. An acceptable level is 15 db below noise (as measured with an oscilloscope or narrow band-pass filter) in the mid-range; with no more than 10 db above the noise level in the upper and lower parts of the audio spectrum.

RF NOISE is defined as the noise due to regeneration in a system employing some RF components, where some of the output signal is induced into some section of the system (generally the input section). Considerable regeneration is possible before the point of instability (where oscillation occurs) is reached. In most cases, RF noise resembles shot or tube noise as it does not exhibit any particular tone. In a station proof-of-performance the difference in RF noise level can be quite large when feeding the antenna system, compared to that obtained when feeding a dummy load.

TRANSIENT NOISE is that noise generally fed into the equipment from the power lines or the grounding system. It is often the result of power systems with high impedance or with insufficient regulation. The tower flasher will frequently show strongly in the noise measurements of the transmitter, where the switching transient will be emphasized in all of the equipment in the circuit - including the test equipment.

FM AND AM NOISE in FM Broadcast Transmitters are specified by the FCC. FM noise is a measurement resulting from a shift in carrier frequency, due to any cause. It is normally measured from a reference of 100% FM modulation with 400 Hz - which is considered to be ±75 KHz carrier swing. AM noise on an FM Transmitter is calculated below an equivalent level of 100% AM modulation of the FM carrier: The carrier level is rectified and measured as a D.C. level at the test point - multiplied by 1.414 to give the equivalent RMS level if the carrier were 100% AM modulated - and the AM noise measured and compared to this simulated level.

TYPICAL SYSTEM - Figure 7 shows a typical system in a broadcasting station with the associated signal levels and the corresponding noise levels. The system is all inclusive from the microphone input to antenna output. The upper shaded area represents the signal range, the lower textured area shows the noise level above −130 dbm, and the plain area in between shows the margin of noise between the relative input level of each equipment and the signal-to-noise established by the first unit in the system.

In a properly designed and operating system, the signal-to-noise is established in the first unit – generally the preamplifier, for a microphone input. Assume that the signal from the microphone is equivalent to −60 dbm, and that the signal generator is delivering this level into the microphone input of the console. The relative input noise of the console and its associated system is considered to be −120 dbm. Thus, the signal-to-noise ratio between the −60 dbm signal level and the equivalent input noise is 60 db. Even though the signal-to-noise of the booster and/or line amplifier is at least 10 db better than the pre-

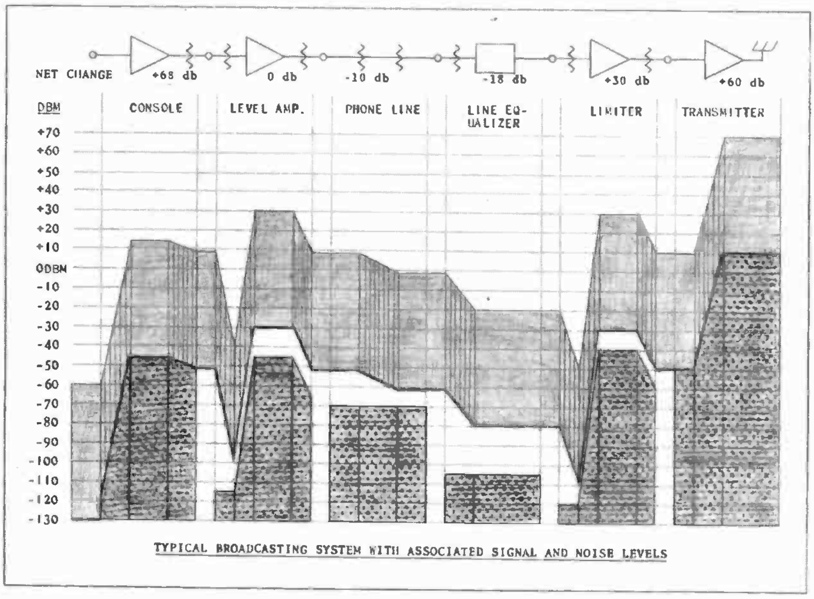


FIGURE 7

amplifier, the original S/N cannot be materially improved in a linear system.

The −60 dbm signal is amplified to +14 dbm by the console, ahead of the 6 db line isolation pad - which reduces it to +8 dbm to feed the "average" level amplifier. The input of the level amplifier is reduced to −40 dbm to prevent distortion in the variable gain stage – then amplified to +30 dbm, the output threshold of compression - and attenuated to +8 dbm to feed into the studio/transmitter telephone line. The line is assigned a residual noise level of −70 dbm; thus, it will not degrade the S/N ratio of the original signal. This typical telephone line has a 10 db loss, delivering a signal level of −2 dbm into the line equalizer at the transmitter site. The equalizer has 18 db insertion loss, giving −20 dbm signal level on its output terminals. It has a residual noise level of −105 dbm, however, and does not degrade the S/N of the original signal either.

The input of a typical limiting amplifier has an attenuator to adjust the incoming signal to the threshold of limiting, generally around −50 dbm. The relative input noise of this point in the limiting amplifier is −120 dbm, which gives a S/N of 70 db to prevent degradation of the original signal. The output level of a typical limiter is +30 dbm, which is attenuated to the transmitter input level of +10 dbm. The S/N ratio of a typical AM transmitter is from 60 to 65 db. Assuming a S/N of 60 db, the transmitter noise will degrade the original signal-to-noise (of 60 db) by as much as 6 db if there is a significant number of coincident noise peaks. A transmitter with a S/N of 65 db would generally degrade the original 60 db S/N by approximately 1 db. A margin of 8 db between the S/N of the original signal or input equipment compared to the following equipments - is considered to be the minimum to prevent any degradation of the original S/N.

SUMMARY - This is essentially a summary of parts of many publications to start with and any further efforts would be redundant. Comments and criticism are invited, especially those with references to standard textbooks normally found in engineering libraries to permit further study. Also, if there is sufficient interest, the NAB may consider some committee action to set up standard ways and means to measure equipment and/or express terms that now have an appreciable degree of ambiguity. Please contact the proper NAB representative and/or the author to show interest in this area.

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