



The Digitally Interfaced Microphone

The last step to a purely audio signal transmission and processing chain.

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With the introduction of a standard defining audio signal transmission, powering, and remote control of microphones with digital output, the question arises, why this step has been taken only now and which features are advantageous for the user.

Analog and digital signal processing: Pros and Cons

Representing signals and data in the digital domain is a necessary prerequisite for allowing mathematically exact processing of signals, i.e., they can be altered, copied, transmitted, and stored in any way. These are well-known advantages.

In contrast, in the analog domain, signal processing can only take place with finite precision, accumulating errors in every step, and without the option of redundant data allowing error correction functions. Error correction functions are very important, for example, when it comes to storing signal information. With CDs, for example, faulty signals caused by the production process can, up to a certain order, be restored with mathematical precision.

In analog signal paths it can thus be said that, in principle, any processing goes along with a deterioration of signal quality, most markedly with a gradual decrease in dynamic range through noise addition and non-linear distortion. Inductive interference from extraneous devices or lines can also pose a problem. Electromagnetic compatibility (EMC) of signal processing equipment is becoming increasingly important, for example, where cell phones are concerned.

After digitizing a signal, however, processing can in principle take place without reduction in quality.

Furthermore, nowadays in the digital domain a vast range of functions can be realized in a compact and cost-effective way, which in the analog processing were difficult to realize, if at all. This is true above all of functions based on intermediate storage of signal data (e.g., FIR filters). In contrast to analog processing, there is virtually no limit to the storage functions available in the digital domain.

So what is the problem that has held up the advent of straight digital "signal processing" for so long?

The answer is that converting an analog input signal into a digital representation is quite difficult, with the conversion itself producing errors and allowing only finite precision and resolution. Above all, however – depending on the extent of the realizable time- and amplitude-based quantization of the signal – a relatively high noise voltage is generated.

The conversion of audio signals, however, poses the highest requirements on resolution, linearity, and dynamic range.

Thus it is no surprise that digitization of audio signals started at the end of the signal processing chain being implemented many years ago, e.g., with the CD, where digital word length is restricted to 16 bits.

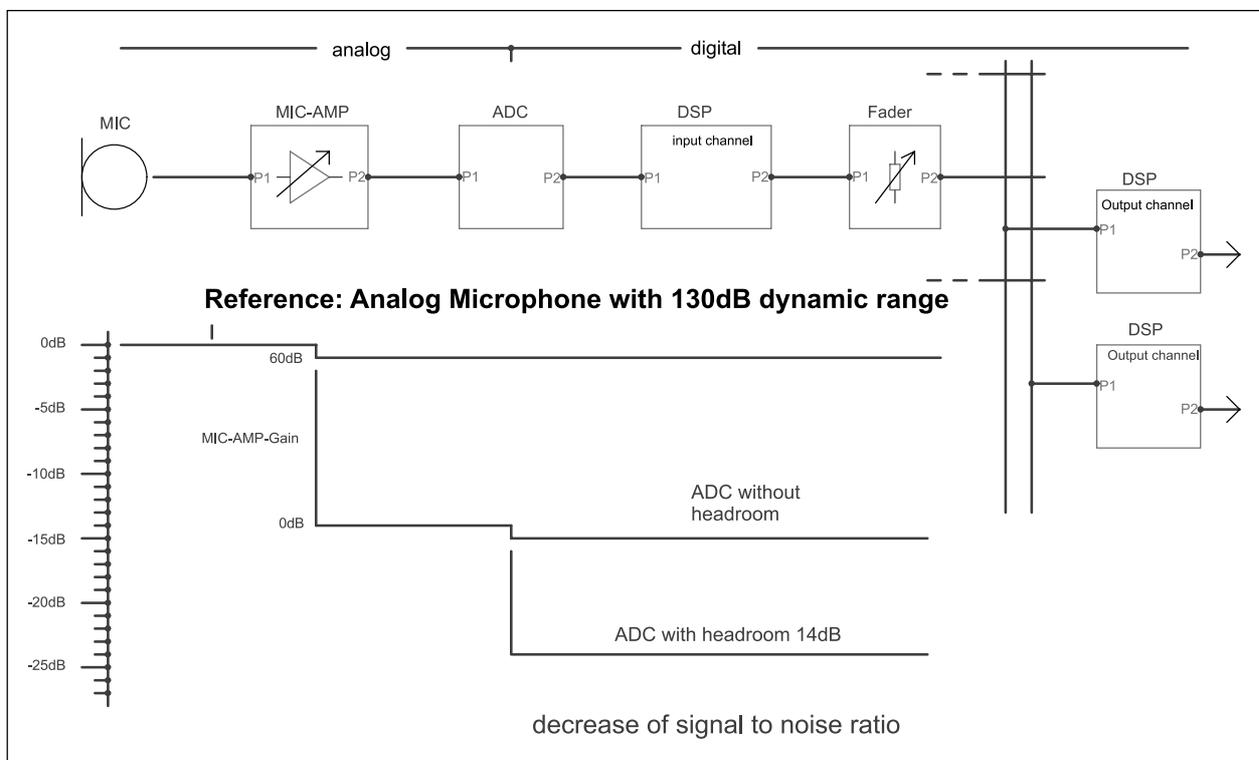
With time, however, analog-to-digital converter technology was improved with regard to word length, linearity, and noise performance, so that today the transition from the analog to the digital domain can take place much closer to the front of the processing path. Thus digitization has come to encompass nearly all components of audio signal processing, such as tape recording, mixing consoles, cross-bar systems, effects processors, etc.

Nevertheless, the available A/D converter-ICs still represent a bottleneck when it comes to processing audio signals. The most advanced Delta-Sigma A/D converters currently available (as ICs) are still limited to a dynamic range of 115 to 120 dB (A-weighted), with a theoretical 24-bit word length.

In comparison, a high-quality analog condenser microphone is capable of yielding a dynamic range of about 130 dB. In order to translate such a dynamic range directly into the digital domain, a much improved analog-to-digital conversion is necessary to avoid noise addition and provide ideal matching of the ADC input voltage range to the signal amplitudes inside the microphone.

One can thus assume that with conversion taking place inside the mixing console or other devices, this will mean having to put up with some signal degradation. Since the A/D conversion there cannot take place until after the necessary level adjustment has been made, dynamics will be affected not only by the characteristics of the microphone preamp and ADC technology used, but also by headroom aspects.

A Typical Transmission Path of Today



In order to process the microphone signal, the microphone output voltages must be matched to the input voltage range of the A/D converter or the console level. This inevitably takes place by means of an ordinary analog preamp. Amplifiers of this type always influence signal quality considerably.



This is because even the best microphone preamps available attain self-noise levels as low as those required only when set for maximum gain. In practice, this will seldom be the case, especially since condenser microphones have a relatively high output level. Moreover, such amplifiers can cause problems because of their sensitivity to electromagnetic interference on the input lines.

The available ADC systems constitute an additional bottleneck with regard to the achievable dynamic range. This becomes especially apparent if one pays attention to maintaining the general headroom necessary to avoid any brief clipping. (See fig.1 "Decrease signal to noise ratio".)

Advantages of a Digitized Microphone Signal

Not mentioning other advantages with regard to expanded functionality, it is in the interest of any manufacturer of high-quality microphones to retain full control over the quality of the digitized microphone signal.

The technical objective is defined so that high-quality digitization of the signal output by the capsule occurs right in the microphone (in the first step of processing, as it were), and level adjustment or other processing steps only occur once the signal is in the digital domain. This provides ideal conditions for preserving the quality of the signal coming from the microphone.

Realizing this objective, however, also permits complete elimination of expensive analog preamps and A/D conversion in the mixing console. This is not only an advantage when it comes to signal quality, but will probably also result in considerable cost savings.

To achieve this objective, Neumann has developed a technique (for which the company has applied for patent protection) that permits almost perfect conversion into the digital domain, retaining practically the full dynamic range and fidelity of analog microphones. The output signal from the microphone capsule is processed without any additional amplifiers between the microphone and the digital domain.

In a procedure similar to known gain-staging procedures, two separate conversion circuits (generally in one IC) are employed. In contrast to those procedures, however, neither signal path is overmodulated. The extremely critical switching processes in the signal are avoided completely. Moreover, both converters are used to represent the useful signal up to the maximum signal level (of the overall system), thus ensuring maximum resolution at any signal level.

The result is an internal digital 28-bit signal with a dynamic range of 133 dB with a "source impedance" of 140 pF and 130 dB (A-weighted) with a microphone capsule. The noise of the digitization process is equivalent to a dynamic range of 140 dB. This means that any noise of the digitizing process is largely masked by the analog noise of the mic capsule.

The Gain-Staging Method

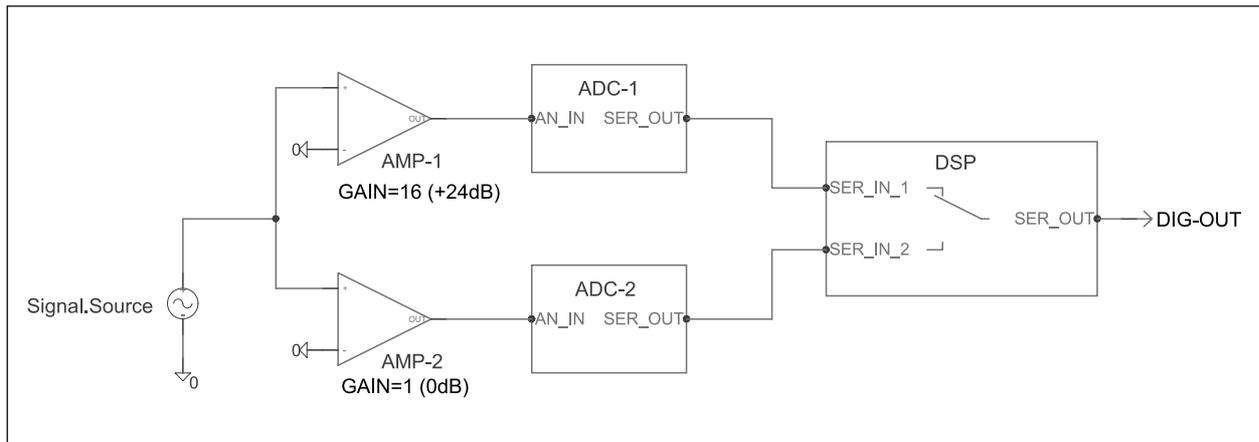


Figure 2 illustrates the basic mode of functioning of a gain-staging method:

Here we see two signal paths that are fed by the same input signal. The first, upper signal path incorporates an amplifier with a fixed amplification of, for example, 24 dB (a factor of 16) followed by an A/D converter. The lower signal path is similarly designed, except that here the amplifier only has an amplification of 0 dB (a factor of 1). The digital output signal is formed in the subsequent digital signal processing by passing through either the signal from the first signal path or the signal from the second signal path as the output signal. The signal from the first signal path is used until the amplitude of the amplified signal is clipped by the A/D converter. The DSP automatically detects this kind of clipping and switches immediately to the second signal path, which carries the unamplified and thus unclipped signal.

The noise gain for increasing overall dynamic range occurs because the signal from the first signal path in the DSP must first be attenuated by the amount of the amplification in amplifier 1 before it is sent to the output. This similarly reduces the noise of the first A/D converter.

Thus the output signal consists of two partial signals that must be matched precisely so that switching from one signal to the other results in a flawless output signal.

Signal Behavior in the Gain-Staging Method

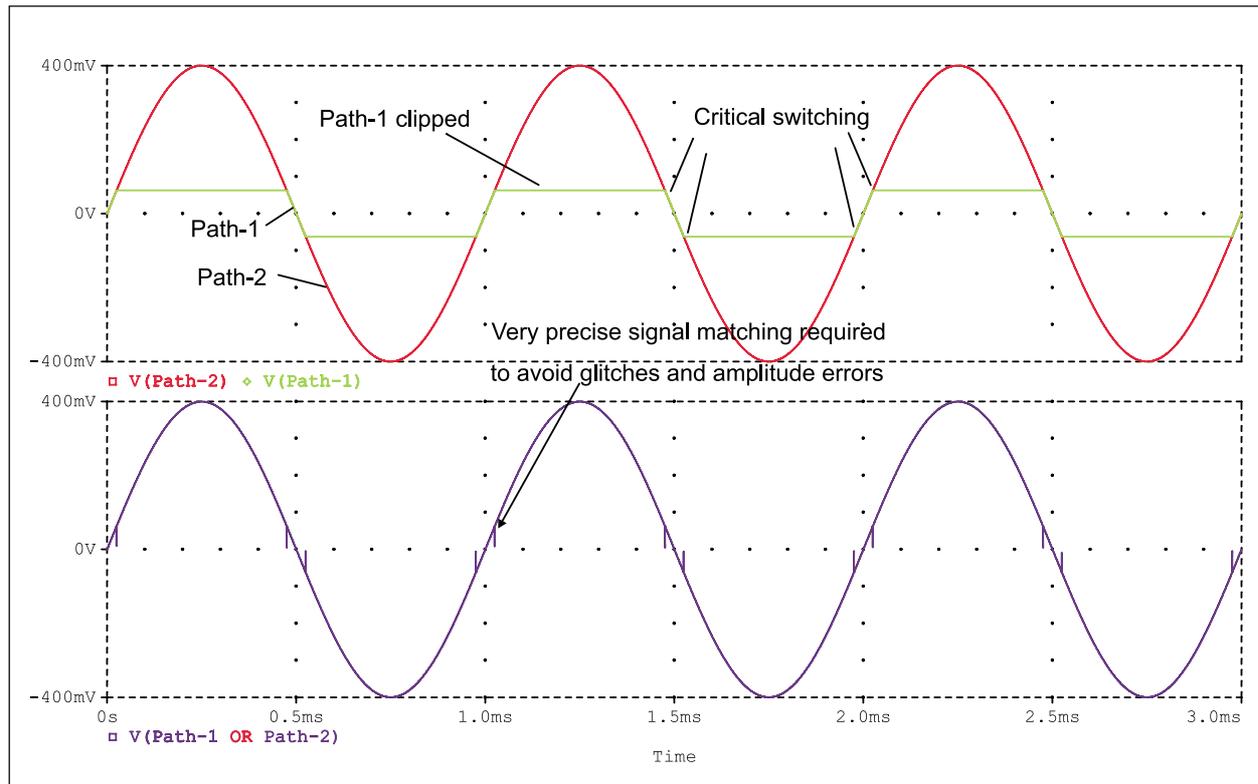


Figure 3 is an idealized representation of the signals in the gain-staging method:

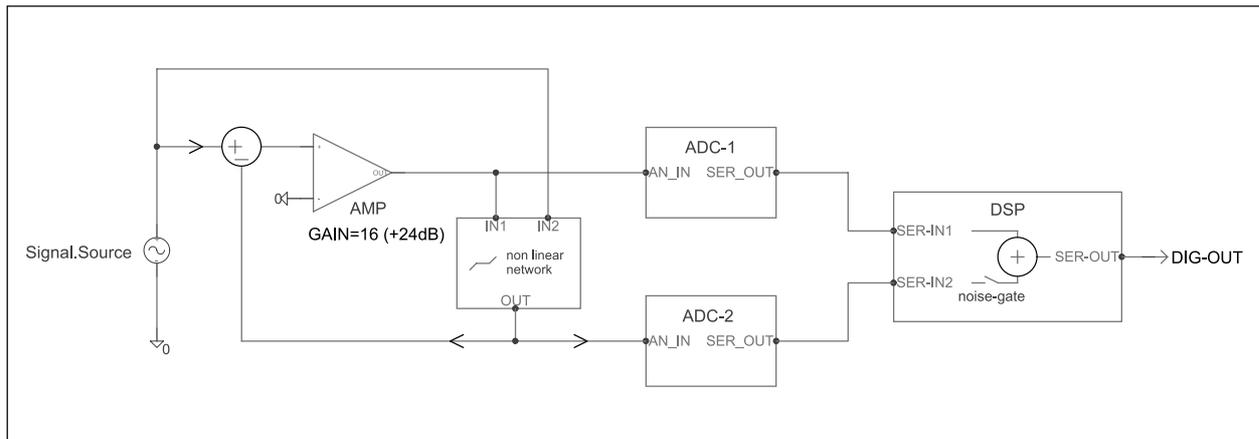
Signal 1 (PATH 1) is generated by the overmodulated A/D converter 1, which is converting the amplified signal that is attenuated (together with the noise components of the converter) in the DSP by the amount of the amplification of the first signal path. Signal 2 (PATH 2) of A/D converter 2 represents the unamplified signal of the second signal path, which is used to transmit high signal amplitudes.

Thus the output signal consists of signal components from both signal paths, which, taken individually, constitute highly distorted signal components. Only absolutely precise combination of both signal components leads again to an undistorted output signal. This refers both to the amplitude ratio of the signal components, depending on the finite amplification precision of the first signal path, as well as to the absolute chronological correlation of the two signals. Even a time error in the nanosecond range can cause undesired distortion products. The curve in the lower diagram shows the combined output signal, in which the effects of even small matching errors become visible.

For this reason, the switchover point from one signal path to the other is extremely critical. Above all, the problem is that filtering of the type caused by the system (decimation filtering) when applied to the digitized signals in the A/D converter generates invalid information in the filter of the overmodulated signal path. This means that switching between the signal paths cannot occur ideally. Above all, this affects signals on the top end of the audio-frequency transmission range.

There are gain-staging converters that use intricate software algorithms in the subsequent signal processing solely to minimize these effects that are inherent to the gain-staging method. For example, the difference in amplification between the signal paths is determined and taken into account through computation when combining the signals, or switching back from the second to the first path is delayed until the conditions for valid data have been fulfilled in the first path again.

The Neumann A/D Converter



The method developed by Neumann employs an entirely different mode of operation.

Figure 4 is a schematic diagram of this converter technology:

Here, too, we find a first signal path with an amplifier that has an amplification of, for example, 24 dB (a factor of 16) followed by a first A/D converter. Then, there is a second A/D converter. The input of this second converter is fed by the output signal of a non-linear network. The input of the non-linear network is connected alternatively to the output of the signal source or the output of the amplifier. Now the non-linear network has a transmission character that does not permit low signal amplitudes to pass through. These are signal amplitudes that are still significantly (6-10 dB) below the clipping point of the first A/D converter. At higher signal amplitudes, the non-linear network becomes increasingly conductive, resulting in an intermediate signal that is subtracted from the input signal of the signal source (linear) on the one hand, while on the other being digitized via the second A/D converter. This process results in a signal, at the output of the amplifier and the input of the first A/D converter, that is defined by the difference between the linear useful signal and a non-linear intermediate signal. This signal has, in effect, the character of a compressed amplitude curve, so that the first A/D converter does not clip, despite the amplification set in the first signal path all the way up to the full-scale limit of the overall system.

The second A/D converter now translates the non-linear intermediate signal into the digital domain. In the DSP, first the signal of the first A/D converter (incl. noise) is attenuated by the amplification factor and then added to the non-linear intermediate signal carried by the second A/D converter. Since distortion of the two signals is always exactly complementary, an absolutely clean digital output signal results.

This is also completely independent of execution and tolerances of the non-linear network, since what is added in the DSP is always only that which was subtracted linearly at the input of the analog amplifier.

The output signal is always formed from the addition of the two paths. Similarly to a noise gate, the detrimental (because it has not been attenuated) noise of the second A/D converter is always switched off when the second signal path contains no signal.

Signal forms in the Neumann converter

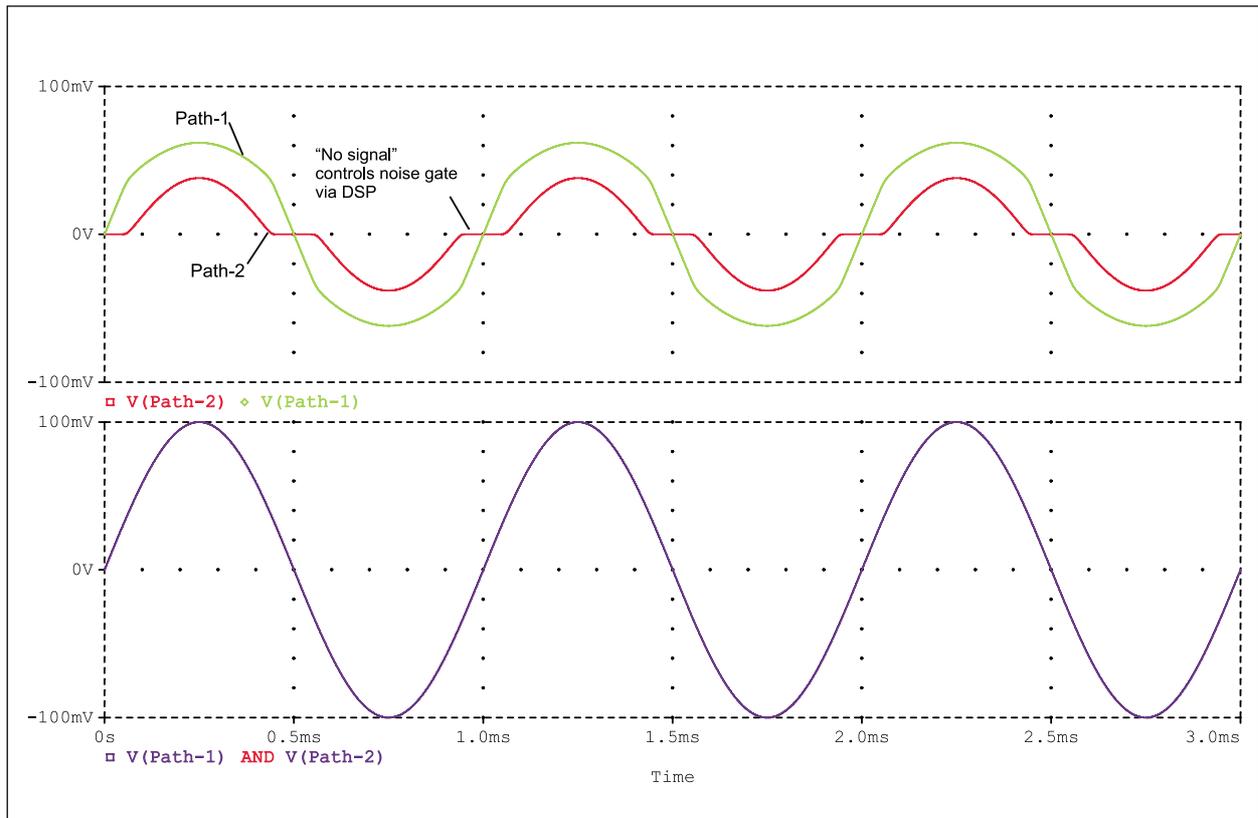


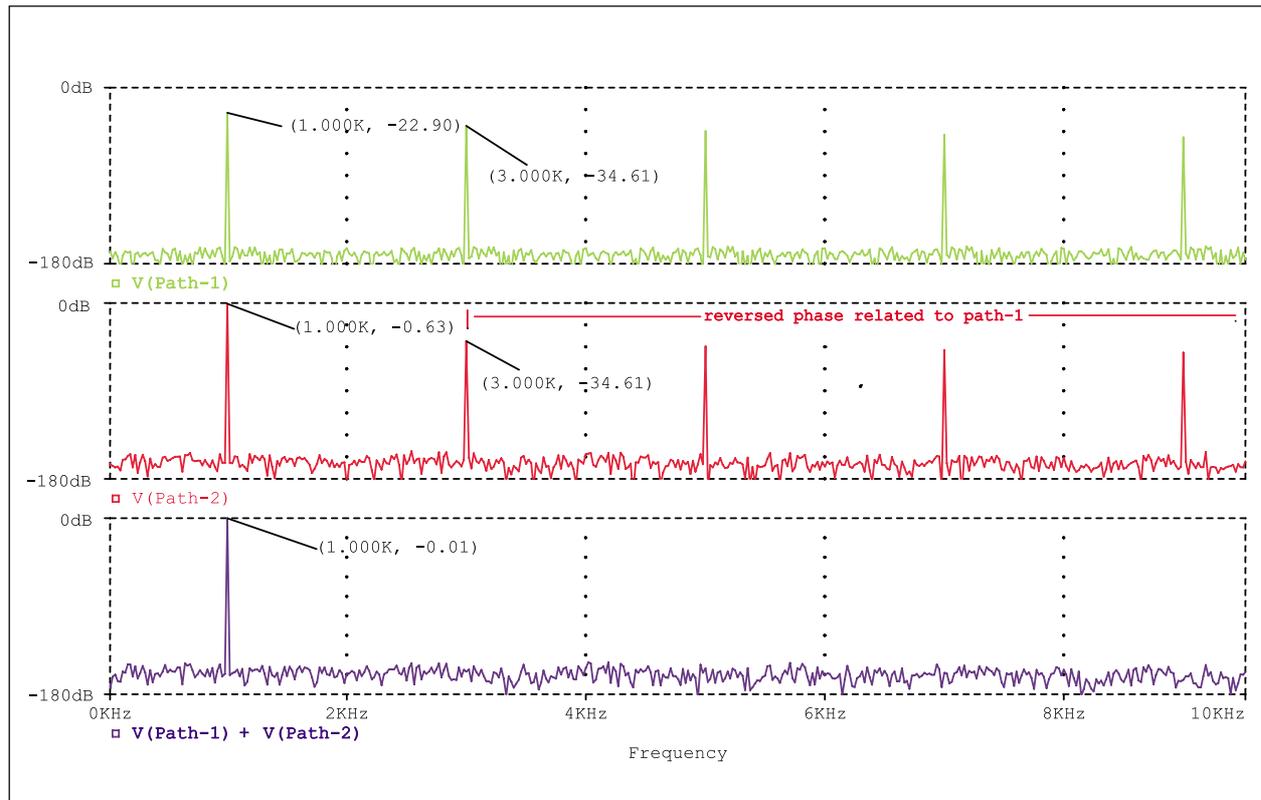
Figure 5 regarding the amplitudes of the signals should illustrate this once again:

The two upper diagrams show the complementarily distorted signals before straight addition in the DSP. The bottom diagram shows the digital output signal after addition of the partial signals.

It is clear from the amplitude curve of the intermediate signal that no signal is formed up until a specific input level. In practice, this value is approx. 30 dB below the full-scale limit. In other words, at lower levels – similarly to the gain-staging method – only (undistorted) signal components are transmitted via the first path. At higher levels, the output signal is always generated through addition of both signal paths.

The non-attenuated noise components of the second A/D converter are always switched off if no signal is being transmitted in the second path. This makes this switching operation completely non-critical and means that it has no influence on formation of the output signal.

Figure 6 shows a Fourier analysis of the signal components:



The upper two diagrams show the two signal paths before they are combined in the DSP. Note that (while this is not generally represented in a Fourier analysis) the basic frequency that corresponds to the useful information is present with the same phase orientation in both signal paths. The relationship between the amplitudes varies depending on the absolute level of the useful information. The distortion products, however, always occur with the same amount, but phase-inverted.

The straight addition in the DSP for forming the output signal is shown in the bottom diagram. We see that the distortion products are cancelled out completely, while the basic frequency components are added to the signal amplitude of the original source signal.

Characteristics of the Neumann A/D Converter

On the whole, this results in the following characteristics for the Neumann A/D converter:

- Both A/D converter paths are not clipping until full-scale-level
- Parallel operation of both A/D converter paths are working within the top dynamic range effecting the highest possible resolution
- No critical signal switching occurs
- No special DSP algorithms for signal matching needed. There is only simple signal addition used and noise gate function for signal path 2
- The internal resolution is 28 bit
- The A-weighted dynamic range is 133 dB with the real microphone capsule, and 140 dB with A/D converter input shorted



This step opened up the way for development of a first digital microphone that meets the highest demands. The microphone works according to the new AES 42-2001 standard, which specifies the interface between microphones with digital output and the following devices. Development of this standard is largely complete; final adoption is pending.

Simultaneously, the digital microphone demonstrates the new possibilities offered by integration of digital signal processing:

Functions familiar from analog microphones can now be remote controlled.

Moreover, this permits implementation of a host of new remote controlled functions that until now were only available in the mixing console and other subsequent equipment, with the potential benefit being financial as well as technical.

Remote Controllable Functions

First we have the classic microphone switches, like polar pattern, pre-attenuation, and low-cut filter.

Furthermore, digital amplification of the audio signal in 1 dB increments through a range of 0 ... 63 dB is implemented, thus permitting levels matching with any sort of external equipment or console. In contrast to known audio preamplifiers, signal-to-noise ratio remains constant regardless of gain settings. As already mentioned, the costly analog preamplifier and the following ADC inside the console or other device (together with undesired noise addition) can be eliminated.

Certain filter functions, such as capsule equalization or low-cut filters, can be realized more precisely and with fewer side effects in the digital domain. It is also possible to realize additional, maybe customer-specific equalizations of frequency response within the scope of the processing capability available in the microphone.

For the first time, it is possible to take steps to reduce damaging transients in the right place, namely, at the source. This results in a special function: the reduction of signal components that arise only briefly, often only for the duration of half a wave, but with high amplitude – for example, instrument attack, "S"-sounds, etc. Microphones can handle transients of this kind effortlessly, but they often make it necessary to maintain excessive headroom in the subsequent signal path, for example, in order to avoid clipping when recording to a storage medium.

The consequence is poor utilization of the dynamic range available throughout the signal path and the impression of a poorly adjusted or excessively low audio signal transmission.

Examples of other remote controllable functions include muting and signal lights.

AES 42-2001 Standard

With the introduction of the AES 42-2001 standard the audio signal transmission, powering, and remote control of microphones with digital output has been defined.

This paper specifies a vast range of remote control commands which can be implemented or even varied via software download within the scope of the processing capability available in the microphone.

On the other hand, the standard also prescribes the sending of information from the microphone to the receiving device. This is done via the status and user bits defined in the well-known AES/EBU audio data format, which is also used for transmission of the microphone output signal.

Included in the transmitted data are the self-identification including manufacturer name and microphone type, information on the remote controlled functions supported by the specific microphone via the standard interface, and status information such as specific warning functions and readiness for operation.

Microphone Synchronization

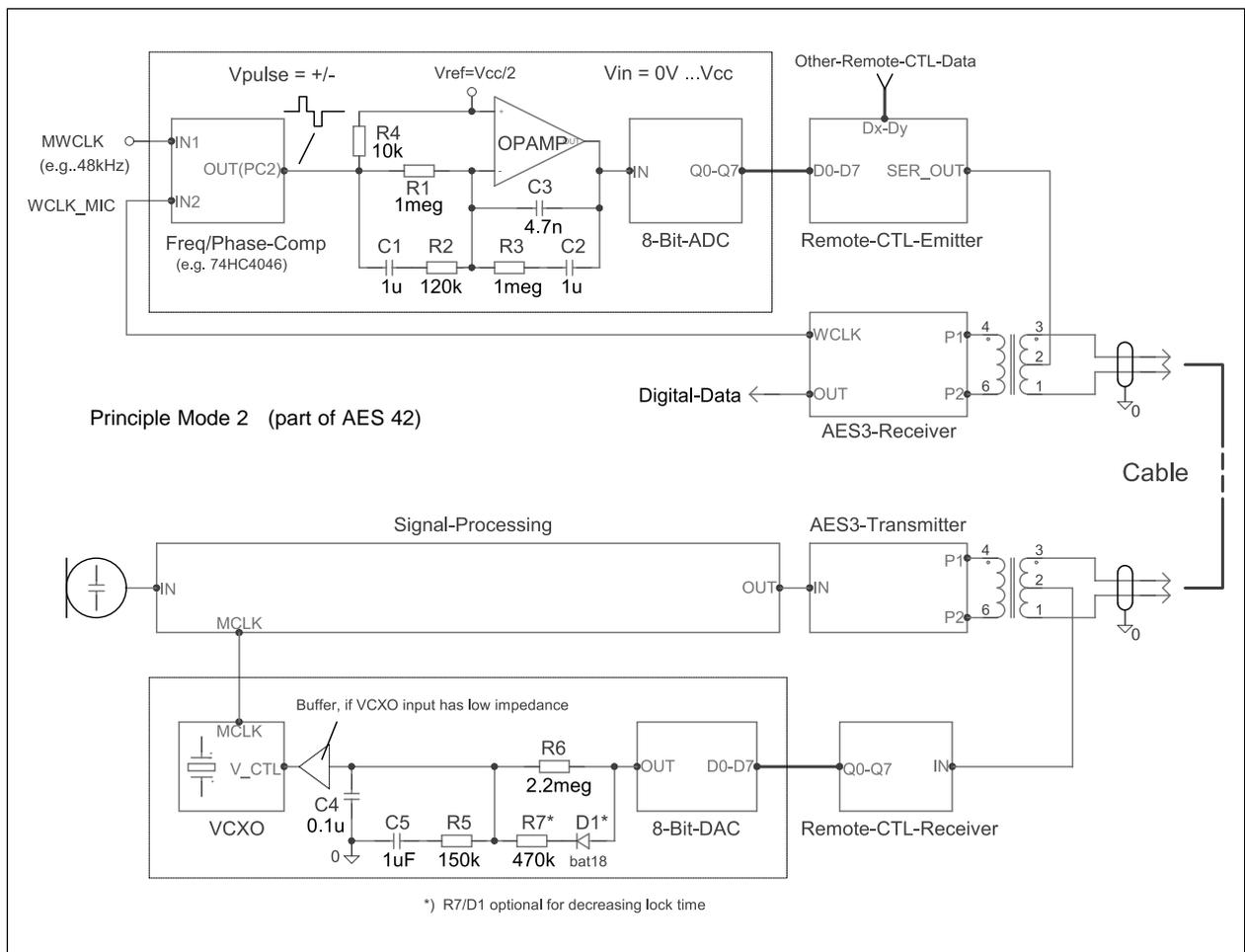
A very specific issue in digital microphone technology is the necessity of synchronizing the output data stream of the microphone with the receiver, e.g., a digital console.

While it is possible to use sample-rate converters (SRCs) for this purpose, the components available today do not fulfill the high standards of quality outlined above and furthermore introduce propagation delays.

Therefore, it was necessary to develop a method of true and stable synchronization of digital microphones, allowing microphone cable lengths of several hundred meters.

Neumann has made a decisive contribution to the solution of this problem by developing a process that has become a fixed part of the AES 42-2001 standard.

The following schematic diagram illustrates the mode of operation:



At the receiving end of the microphone signal (mixing console or a feed device inserted before the console), a master clock performs a frequency-phase comparison. The result is a rather slow feedback signal, used to control a VCXO (Voltage controlled crystal oscillator) inside the microphone. This resembles a closed feedback loop similar in function to the well-known PLL (Phase Locked Loop).



After equalizing and A/D-converting the control signal it becomes an integral component of the remote data stream to the microphone, as specified in the AES 42-2001 standard.

This method is very reliable and produces only negligible jitter amplitudes (in strictly mathematical terms, derived from the control voltage, less than 1 ps in the audio-frequency range).

Conclusion

Summing up, it can be said that the technical foundations for the wide introduction of digital microphone technique have been created.

A comprehensive standard has been compiled specifying digital microphone operation, connection, and control. With the availability of high-quality digital microphones, comparable to the top-range analog microphones, the missing link to a completely digital audio signal processing and transmission path is now available.

The functions that can be implemented in digital microphones will help in allowing realization of high-quality audio signal transmission and recording even with relatively inexpensive, strictly digital equipment.