Continuous digital workflow – a gateway to “high resolution” audio and a higher production Efficiency

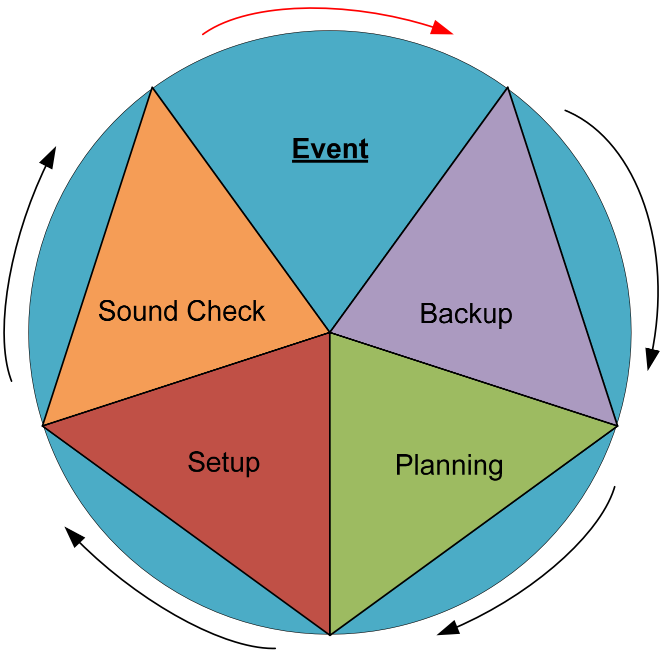
*Wolfgang Fraissinet *, Jürgen Breitlow*,*
*Marcel Babazadeh, Dipl. Tonmeister ***
* Georg Neumann GmbH Berlin, headoffice@neumann.com
** Innovason / LAWO Group, m.babazadeh@innovason.com

1. Introduction

Every PA Event follows the same rules in a repetitive manner. Independent of the type of event (Theatre, Concert, Broadcast or Industrial) the workflow has to adapt to the given time and the allocated budget. For sure - the end result should satisfy or exceed the expectations of the audience.

This cyclical process (workflow) can be divided into 5 parts:

Planning – Setup – Sound check – **Event** – Backup



Each part shall be highlighted and the benefits of using a state-of-the-art digital mixing console which offers full control of digital microphones shall be shown later on. The purpose is to increase the quality of the event and to save time during the cycle.

2. The five parts of the cycle

2.1. Planning

During this stage the technical setup is defined according to the needs of the event, the size and infrastructure of the venue and the availability of equipment.

The number of boxes and cases as well as their physical measures must be defined to inform transportation companies and to reserve space on the venue.

2.2. Setup

The gear will be mounted and powered and cables will be patched. From Microphones to Stage racks to Consoles. Labels and names are given to channels and busses and all necessary Fx and recording equipment will be installed. The lines will be checked and faulty parts will finally be replaced to ensure a smooth Sound check.

2.3. Sound check

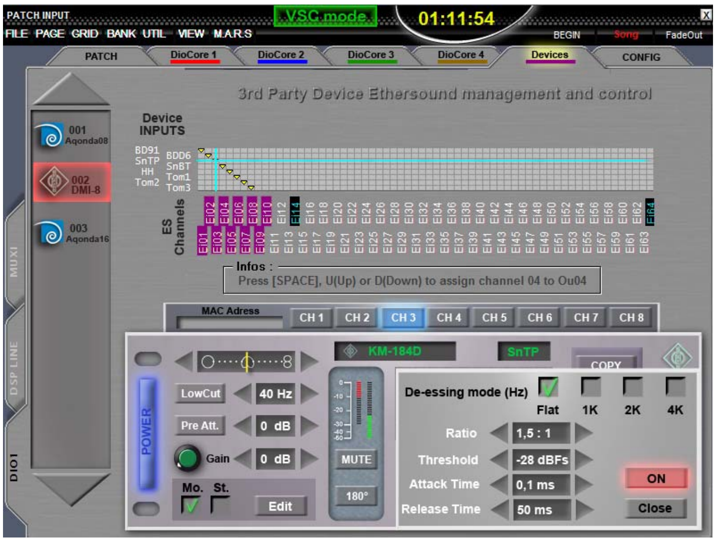
In a very short time window the sound engineer has the possibility to work with the talents on stage. The sound of the main sound system will be adjusted as well as the monitoring system for the talents. In many cases a separate console takes care of the monitor mix.

2.4. Event

During the event the main focus lies on operation of the sound sources from the console and the handling of the mix for the sound system or transmission and distribution of various sub mixes for multiple purposes. In many cases the event is archived/ recorded for latter usage (CD/ DVD publication).

2.5. Backup

After the event is finished everything is put back in its place to be ready for the next event – be it the upcoming theatre rehearsal or the next gig in a city far away.



3. The benefits

A high class digital mixer offers a large amount of **flexibility for the planner**. On a small physical footprint it is possible to handle vast numbers of sound sources. The surface layout can accommodate to the specific needs of the event, so any fader can have any function and a set of user defined panels, knobs and switches ensures direct access to the desired function. Since all FX can be used from within the console **the amount of outboard gear is dramatically reduced** and the overall weight and packing space is very small.

Using digital microphones will deliver a constant and reproducible sound image of the highest class even offering his own dynamic processing which adds up to the signal path of the system. Working with well established audio networks like EtherSound enables the planner to flexibly distribute sound sources within the venue and beyond without vast amounts of additional equipment.

It also allows **clever redundancy concepts** which are vital, especially for industrial events.

During the setup the technicians benefit from the fact that a cable with a digital signal does not suffer from external interference. Since the signal will be digital right after the capsule there is no need for fear.

A built in signal generator allows for a quick line check to see if the connections are working. It can be turned on and off from the mixing console.

All attenuation switches for the microphone are now found on the surface of the console and their settings can therefore be edited and saved in the mixing desk as part of the project file.

The clock source is taken from the network, so **each item works in a fully synchronous, phase coherent mode** just by one connection (important for AES 42 Mode 2).

Using small stage racks allows elegant and short cabling for the microphones. The connection between the boxes and the mixing console are lightweight and use digital multichannel standards (e.g. 64 channels bidirectional on a CAT 5 cable). A switchable, monochrome LED on the microphone eases the allocation on stage.

The whole setup of the desk can be prepared offline and loaded on any system during the setup and can be stored on external or internal memories.

During the Sound check the sound engineer profits from the **huge dynamic range of the digital microphones** which can virtually not clip. He can share the sources in the audio network by leaving the Microphone gain untouched and adjusting his own digital trim, Therefore all members of the network will have the possibility to share the sources without the need of any local gain compensation (e.g. Monitor Mix, FOH Mix, Broadcast Mix).

Modern consoles offer the Virtual sound check (VSC) as part of their system. It is then possible to record an excerpt of the performance and replay the tracks on their own channels of the console even after the talents have left the venue. This saves time and nerves of everyone involved – leading to the fact that **the engineer can adjust the sound with much more care** and talents can focus on their performance.

Extensive Snapshot functions including all parameters of the consoles and the microphones allow very distinct and different settings to change rapidly and predictably.

Everyone will be astonished of the **absence of audible noise** within a digital system. During the event the overall **sound quality and dynamic range are improved**.

In the same time the built-in recording system can record each track separately on a hard disk and offer it to a postproduction facility afterwards.

All saved parameters can be recalled and previewed with a button press and if necessary they can be individually isolated, removed or overwritten during operation.

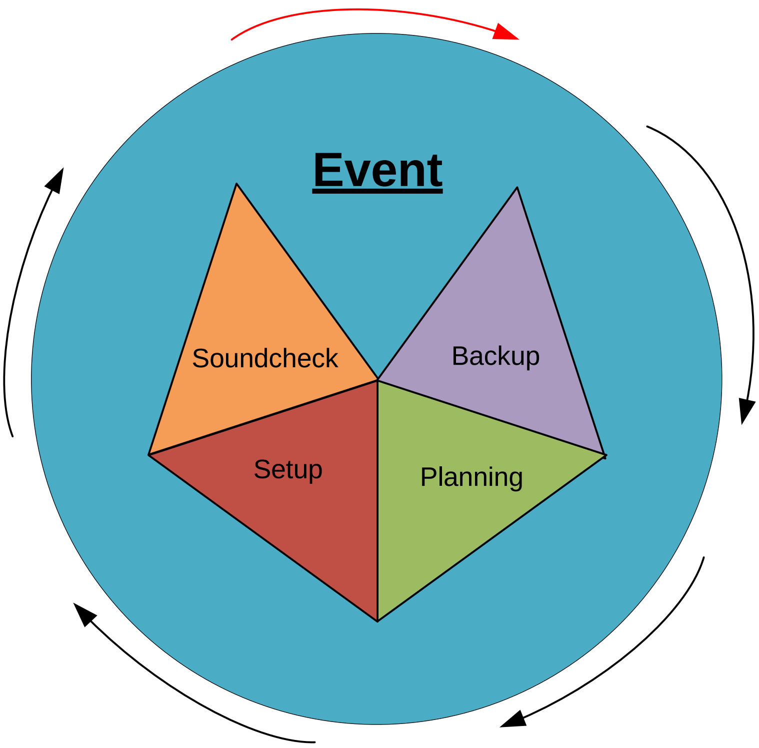
Since all parameters are stored within the project file of the mixing desk, the microphones can be taken off their stands after the event without paying attention to their settings. Having reduced the amount of cabling and external gear by using a high class digital mixer the backup time will be reduced remarkably.

4. Conclusions

By reducing error sources (e.g. interference, analog cabling) and keeping many known work steps (e.g. xlr cables for microphones, stageracks) **it is possible to improve the sound on stage without changing the existing workflow dramatically**.

Working with a modern console that supports all the parameters of digital microphones based on AES 42 allows an **improvement of the actual workflow**.

The time which is needed during the cycle to prepare the event can be reduced, so that the focus on the event itself is stronger and **the end result gets noticeably better**.



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Abstract

We used the occasion of a recording classical music with the Colorado Symphony Orchestra in Denver with digital microphones only to think about the changes in workflow related to digital microphones and their integration into digital consoles. A recording with digital microphones is described together with the setup und workflow. The technical background of the advantages of digital microphones is presented. The current status of integration into mixing desks is described with its impact on the workflow. Finally it is shown how a complete integration of digital microphones improves the workflow in live event environments.

1. The recording

We just completed an all digital recording of classical music composed by Brahms, Mozart, Beethoven and others with the Colorado Symphony Orchestra using the new Neumann KM 133 D as an array of main microphones in 5.1 surround. Wolfgang Fraissinet served as the producer for these recordings. We have used several digital microphones from the Neumann Solution-D series together with various Sennheiser MKH digital microphones. The new diffuse-field-equalized Neumann miniature microphone KM 133 D was used in a front line-up covering the acoustic depth, the audible space and for clear location of every instrument on stage.

Both lines were used in a synergetic combination for various spot micings within the orchestra (woodwinds, two harps etc.). The rear channels have been covered with a digitized Neumann dummy head utilizing a Sennheiser MZD8000 as a digital interface. All these digital mics have been operated on the same platform of accessories (Neumann DMI 8 = digital microphone interfaces for 8 channels).

The artistic performance of the pieces has been well balanced by the producer in cooperation with the chief conductor and the entire orchestra. The result is a new CD with tastefully arranged pieces of well known classic music. It represents a new state of the art digital recording which will be released in Q1 2011.

2. Advantages of digital microphones

There are several advantages of digital microphones which make them a favorite choice for the sound engineer. They improve the quality of sound on one hand. On the other hand the workflow gets simpler and they help to save time during the recording.

2.1. Dynamic range and gain

There are two aspects which need to be addressed when talking about digital microphones. The overall dynamic range of digital microphones is optimal compared to analog microphones with a separate AD converter because the AD conversion is fully adopted to the voltage and impedance requirements of the capsule.

Figure 1 shows what SPL range is covered by a standard miniature microphone. The lower end is 13 dB. At the top end it is 138 dB. When this is connected to a pre-amp and AD converter with minimum gain the maximum voltage coming from the microphone fits quite well to the maximum allowed voltage of the pre-amp/AD converter. But at the lower end there are about 12 dB lost because of electronic noise in the pre-amp. To compensate the loss at the lower end gain has to be increased which results in a much better self noise at the lower end. But this gain reduces at the same time dramatically the headroom at the top end. In the middle of figure 1 the effect is visible: At 64 dB gain the equivalent self noise of the whole signal chain reaches the numbers of the microphone by 1 dB - but at the same time the maximum SPL is reduced by 64dB.

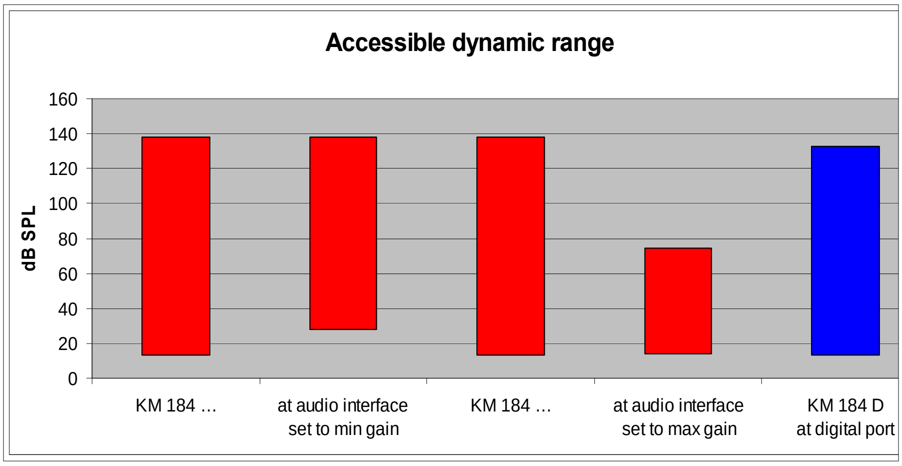

Figure 1: Dynamic range in analog and digital signal chains

For the digital microphone on the right side of figure 1 the AD conversion fits so well with the capsule that it captures the whole dynamic range at once.

The conclusion from figure one is that with digital microphones there is no need the think about gain in the signal chain to improve sound quality. Gain is only needed for level matching it does not change audio quality or noise.

The second aspect of the dynamic range is the self noise of the whole signal chain and the recording. With digital microphones there is no noise added later in the chain which results in an audible lower noise floor in the recording.

2.2. Remote control features and internal DSP

Digital microphones offer different features which can be remotely controlled. Features like polar pattern, pre-attenuation and filter are typical for microphones. Other features like phase reverse, gain, compressor/limiter and peak-limiter may now be part of the microphone and are typical for console channel strips. With digital microphones all these features can now be set remotely and there is no need to set any switches at the microphone anymore. This offers for the first time the chance for a real total recall of parameters for a recording session.

2.3. Workflow and integration

There are currently three levels for the integration of digital microphones into mixing desks available on the market.

Level 1: The starting point for using digital microphones is an AES3 input with digital phantom power and a sample rate converter. All benefits regarding the dynamic range of digital microphones can be used in such a setup. There is no remote control available and the sample rate conversion will increase the latency of the signal chain. But in a many cases this setup will already fully fit the user's demands. The main limits are the missing access to polar pattern and pre-attenuation because the lack of these features cannot be compensated afterwards.

Examples for the first level of integration are SoundDevices 788 or input cards of Digico's stage boxes.

If a microphone offers DSP functions the DSP will work but parameters cannot be changed. In some cases with un-experienced users it might even be desired to use DSP function without a possibility to change parameters by the user.

Level 2: In the second level of integration digital microphones are powered and some of the features may be changed. An example for this is the Stagetec Nexus system where only microphone typical parameters may be changed. All other functions like compressor or limiter would be used in the mixing desk where they are available as well. For the user it might even be favorable to use DSP function is the mixing desk because he wants to have the same DSP functions for all channels.

Level 3: The highest level of integration allows the control of all features in the microphone. Innovason's integration of Neumann DMI-8s into their eclipse mixing desks is an example for this. With synchronization of digital microphones (mode 2 in AES 42) this setup allows minimum latency because there is no latency added from sample rate conversion.

All three levels offer benefits to the user - but as far as we can see the seamless integration of digital microphones only offers a really easy workflow.