

# CHALLENGES OF TESTING DIGITAL AND WIRELESS AUDIO DEVICES

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## INTRODUCTION

Mobile devices (Bluetooth headphones, smartphones, tablets, ...) are extremely relevant because almost everyone is carrying at least one device in their pocket. Car infotainment systems and fixed sound installations (alarm and public address systems) accompany everyone's daily life. Yet, those audio systems hold specific challenges when the audio reproduction needs to be measured (especially in quality assurance where defect symptoms need to be detected sensitively and quickly while measuring under tough acoustical conditions).

For those audio systems, the following measurements are recommended for quality assurance:

- Frequency response
- Phase response and polarity
- Harmonic distortion
- Sensitive Rub&Buzz for detection of
  - Loudspeaker defects
  - Assembly faults (e.g. loose screws and parasitic vibrations)
- Air leakage detection (e.g. due to mounting or gluing faults)

Ambient noise detection or immunity is strongly recommended to avoid false rejects due to ambient noise.

Sophisticated signal processing algorithms need to be fed with relevant response data from the device under test. Due to significant delays, feeding correct data is quite challenging when using external signal sources. This article discusses methods for coping with unknown, varying or even unpredictable delays in the measurement chain while still fulfilling all requirements for quality assurance applications.

## THE ANALYSIS WINDOW

The analysis window describes the time window where response data is recorded and fed to the analysis algorithms. It usually has the length of the stimulus.

Any audio measurement system needs to obtain the response of a device under test. In standard measurement situations (see Figure 1), the stimulus is played back without a delay and the capture process is synchronized to the playback. Thus, the temporal position of the response data is known and the analysis window is optimally placed automatically.

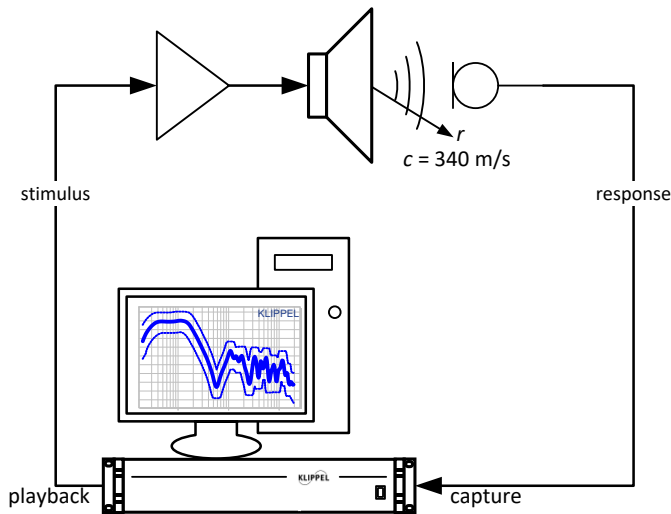


Figure 1 Standard acoustic measurement setup with synchronized playback and capture

When one measurement device is used, playback and capture processes are sample-synchronous and the internal delay is compensated automatically. In that case, the response is usually delayed only due to distance of acoustical source and microphone. The acoustical delay causes a slight shift of response data in the analysis window (see Figure 2). The shift causes the last part of the response to be outside of the analysis window. As the delay is usually very small (see Table 1), the number of missed samples is negligible compared to the complete response.

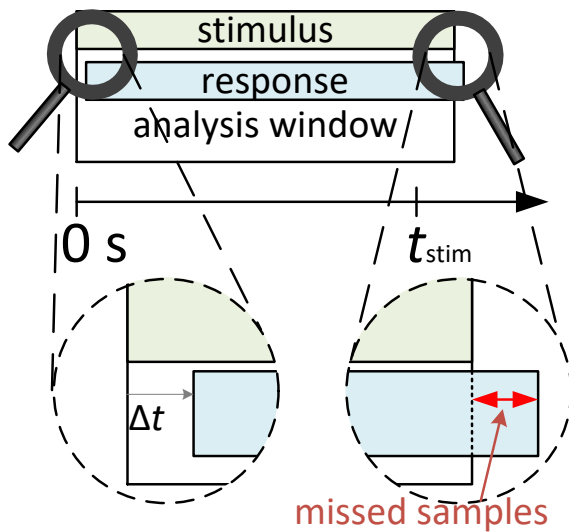


Figure 2 Illustration of response delay in standard measurement setups

For a single sweep, the relevance of the missed response samples depends on the actual delay  $\Delta t$  and the length of the analysis window. For measurement setups with up to 1 m distance between source and microphone, the impact is still small for a 1 s sweep. For ultra-fast stimuli, distances above 1 m should be avoided.

Table 1 Relevance of delay in standard measurement setups

Distance $r$	Delay $\Delta t$	Missed response data	
		0.2 s sweep	1 s sweep
0.1 m	0.29 ms	0.15 %	0.03 %
1 m	2.94 ms	1.47 %	0.29 %
10 m	29.41 ms	14.71 %	2.94 %

## WHAT IF DELAYS ARE NOT NEGLIGIBLE?

In some measurement setups, the distance between acoustical source and microphone is not the only reason for a delay. Other factors may introduce significant additional delays.

Examples:

- Usage of 3<sup>rd</sup> party audio devices for playback and/or capture processes instead of a measurement instrument (see Figure 3):  
Drivers and the operation system's audio architecture will introduce latencies. Due to dynamic definition of the driver's buffer sizes, those delays will be different on every driver initialization (see Figure 4).
- Sample drift of audio devices: Using mixed configurations (different audio devices for playback and capture) without synchronizing sampling clocks will cause a drifting delay versus time.

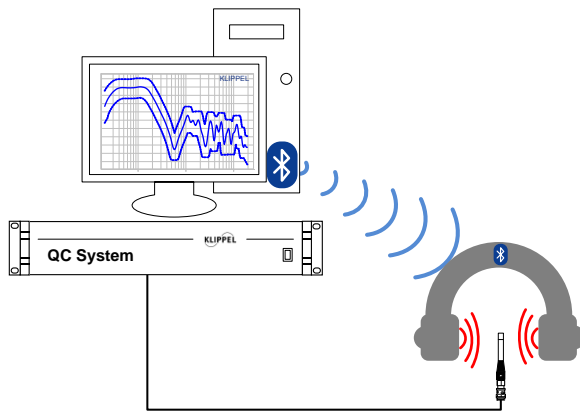


Figure 3 Application example: Using Bluetooth headphones for audio playback

- Other delays in the signal chain: If digital signal processors and wireless technologies are included in the measurement, additional delays will be introduced. Those delays are usually constant.

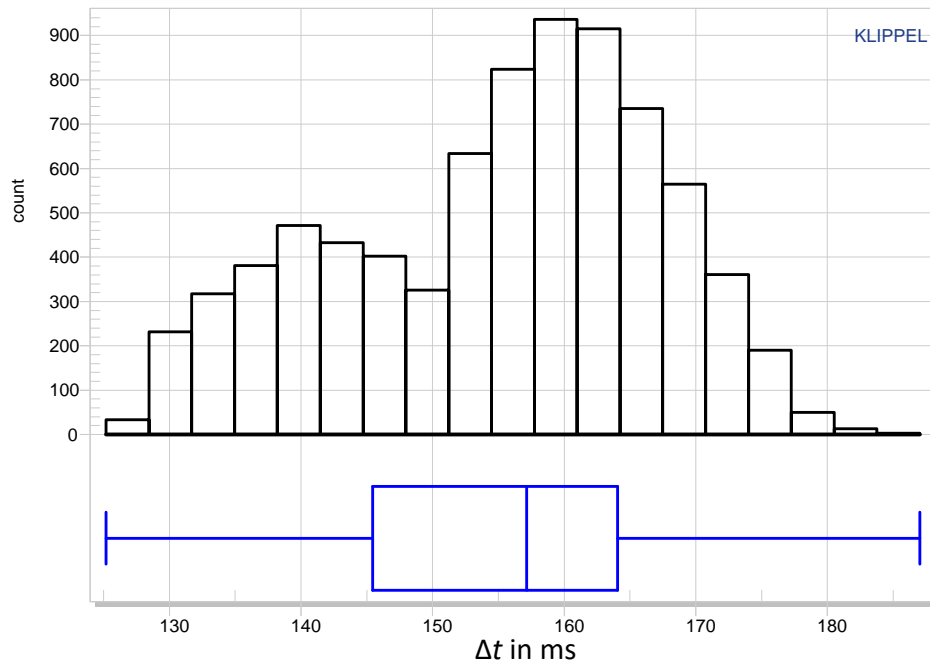


Figure 4 Distribution of delay caused by reinitialization of audio devices for setup as shown in Figure 3

Figure 4 shows the delay distribution of 7800 measurements using a mixed configuration (playback with *Bluetooth* audio, capture with *Klippel QC* hardware). The audio devices were reinitialized before every measurement. The mean delay of around 155 ms varies by approximately  $\pm 35$  ms. (That would correspond to an acoustic delay due to a varying distance of  $\pm 12$  m.)

The delay and its variation cannot be neglected anymore. Otherwise a tremendous amount of data is lost (see Figure 5) for the analysis yielding measurement results with questionable relevance.

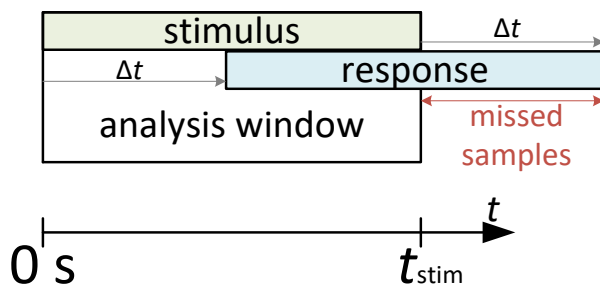


Figure 5 Illustration of huge data loss due to significant delay

## WHAT IF THE MOMENT OF PLAYBACK IS COMPLETELY UNKNOWN?

In some measurement setups, the moment of playback (and thus the moment of time when the response hits the microphone) is completely unknown. This is the case if playback and/or capture processes are completely

autonomous and not synchronized at all. An example is the manual trigger of playback of a sound file on a tablet device (Figure 6). Other examples are complex audio systems, where the playback is triggered by automation processes (e.g. in-situ quality control in automobiles at the end of the assembly line) or mobile measurements (e.g. checking the loudspeakers of an alarm system in an airport with a handheld recorder and post-processing the response data).

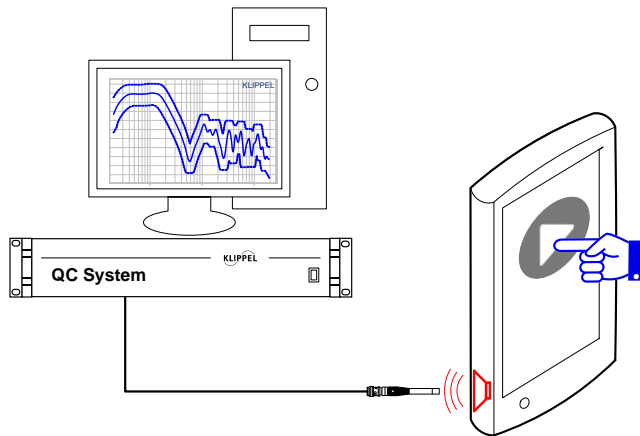


Figure 6 Application example: Using autonomous playback when testing tablets or smartphones

These wave file based measurement setups have to be used if the audio system under test does not provide a suitable signal input/output. The problem of placing the analysis window in the correct moment of time is similar to the mixed configuration setup, but with unpredictable delay between measurement start and actual playback.

## SOLVING THE PROBLEM

### STATE OF THE ART

Measurement systems that cope with unknown/varying or even unpredictable delays often use one of the following solutions:

#### INCREASE LENGTH OF ANALYSIS WINDOW

By using a longer analysis window, the need to position the initial analysis window perfectly is relaxed, assuming that the complete response is captured somewhere in the large analysis window. Subsequent algorithms (e.g. the calculation of impulse response) may place a refined analysis window of shorter length at a more precise moment

of time.

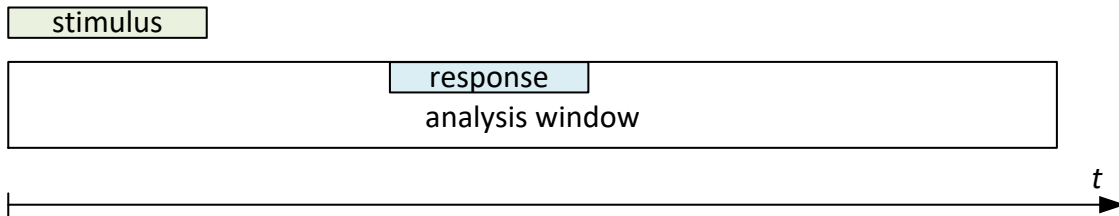


Figure 7 Large analysis window to handle unknown delays

A major drawback of this method is the time consumed. It is not suitable for automatic wave file based measurements since the assumption that the complete response was captured might not be true. Moreover, this method does not address the problem of sample drift. Even if the complete response was captured, it might not be detectable in the impulse response for fine-tuning due to sample rate deviation.

## LOOPING THE STIMULUS

Looping the playback of the stimulus may eliminate the need to position the analysis window at all. This is applicable for all steady state signals (sinusoidal tone, multitones or noise signals) or even a repeated sine sweep. The analysis window is placed arbitrarily in the response stream. If necessary, the response data may be mapped circularly to yield a corrected response block.

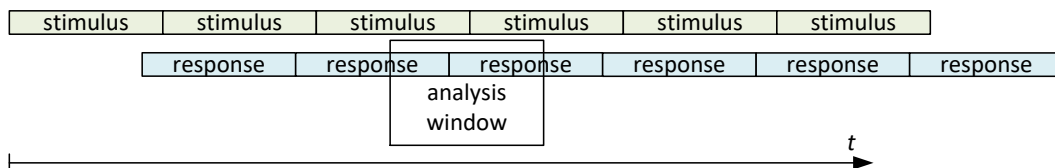


Figure 8 Looped stimulus and response for arbitrary placement of analysis window

This method is obviously very time-consuming and adds the demand of repeatability to the stimulus. Furthermore, a significant amount of energy is put into the system. If a repeated sweep is used, room acoustics might falsify results (e.g. THD) or even mask defect symptoms.

## TRIGGER THE MEASUREMENT SYSTEM

Triggering the measurement system at the correct point of time is the most time-efficient method to position the analysis window. The conventional method uses a sinusoidal energy trigger. This tone is detected in the recorded data stream and as soon as it is recognized, the measurement system starts capturing.

The detection of a tone is quite time-consuming, inflexible and relies on a very narrow frequency band. A sample accurate delay estimation can often not be achieved.

Correlation techniques could also be used to detect the starting point of the response data. Complex stimuli could be used, which provides a more robust method. Usually these methods do not address a possible sample rate deviation between capture and playback devices. Eventually they will fail when the deviation or the stimulus length grows.

## NEW SYNCHRONIZATION METHOD

### SYNCHRONIZATION SIGNAL

The new synchronization method uses either an additional broad-band signal (pink or white noise) or the stimulus itself.

By using an additional trigger signal (see Figure 9), a unique ID (like a fingerprint) is injected in the noise signal. Only this specific ID will be able to successfully synchronize and trigger the system. Thus, the synchronization signal for each test type is unique, which ensures using the correct response for a measurement. An additional noise signal should also be used if very long test stimuli are used for a measurement (above 2 s).

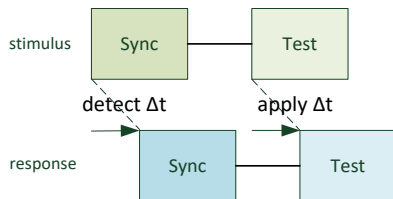


Figure 9 Synchronization with additional trigger

By using the stimulus directly as the trigger (Figure 10), the consumed overall time is kept small because no additional signal needs to be introduced in the sequence.

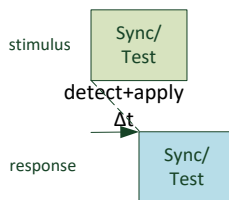


Figure 10 Synchronization with the stimulus as trigger

All synchronization signals provide a sample rate tolerance of up to +/- 10 % for the synchronization process. That means that a sync response is also detectable if the stimulus is mistakenly played back with 44100 Hz sampling frequency instead of 48000 Hz.

### TIME REGIMES

To meet all measurement situations described earlier, two basic time regimes are provided:

1. A dynamic time regime (Figure 11) detects and compensates delays on-line. This mode can be used if the playback and capture streams are under control of the measurement system. A sync response is detected sample-accurate. Subsequent measurements are analyzed with an optimally positioned analysis window. A synchronization

signal is played back before the test signals. Optionally, a resynchronization within the sequence may be used to compensate for drifts of sample rates between capture and playback streams.

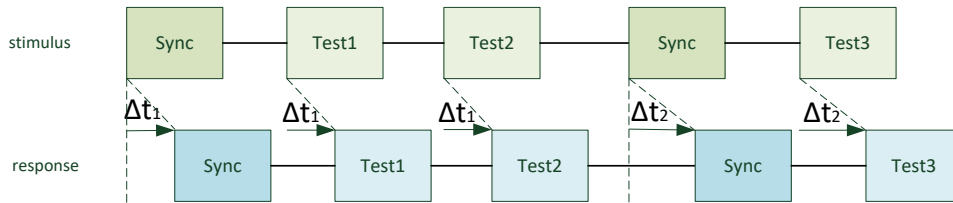


Figure 11 Dynamic time regime with three test signals and a resynchronization

2. The static time regime (Figure 12) handles measurement setups with completely autonomous playback (or capture) processes such as wave file based measurements. Here the absolute durations and gaps between all signals are fixed. Again, a synchronization is requested at the beginning of the sequence to provide optimal placement for the analysis windows of subsequent measurements (now based on an absolute time axis). Optional resynchronization may compensate for sample rate drifts.

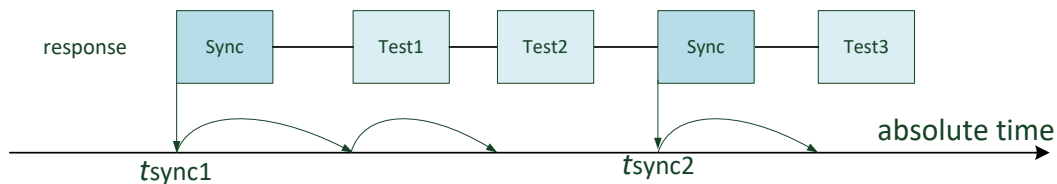


Figure 12 Static time regime example

With the static time regime, it is possible for standard sequences to be reused in several applications (like a standard audio test CD).

## TECHNIQUE

The synchronization search is based on overlapping block-wise analysis (see Figure 13). The evaluation is based on the calculation of the impulse response. If a valid impulse response is found, the synchronization is successful. Based on the position of the main impulse in the window, the delay (dynamic time regime) or the absolute position (static time regime) is determined. The temporal position of subsequent response data is calculated based on the delay or the absolute position of the synchronization.

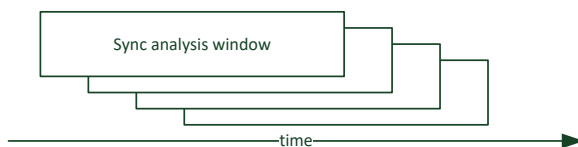


Figure 13 Block-wise processing of the new synchronization technique



When a significant sample rate deviation is present, this technique would fail if used solely. The new method is still able to find the synchronization by reconstructing the original time axis before the calculation of the impulse response.

Multiple paths of sound distribution (and thus multiple peaks in the impulse response) are handled by a smart look-ahead algorithm making sure that always the incident of direct sound is used for synchronization.

## IMPLEMENTATION

This new synchronization technique is available as an add-on for the *Klippel QC* system as the *External Synchronization (SYN)* module. It ensures reliable and efficient synchronization of playback and capture while allowing using the unique features of the *Klippel QC* system.

The setup of the SYN module is as easy as possible. All described features run in the background and do not require user configuration. The user defines a synchronization request only by selecting the channel characteristics from a set of templates. A custom mode provides options for fine-tuning in special cases.

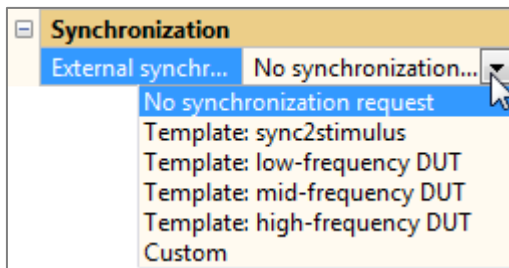


Figure 14 Defining a synchronization with the SYN module in *Klippel QC*

## SOLUTION EXAMPLES FOR SELECTED APPLICATIONS

System under test	Conditions	Solution
<b>Bluetooth headphones</b>	Signal input available (Bluetooth audio system)	Using SYN module in dynamic time regime provides time optimal and flexible measurement sequence.
<b>Smartphones, tablets</b>	Autonomous wave-file based playback (speakers) and capture (microphone)	Using the SYN module in static time regime provides detection of sync response with autonomous playback. Captured wave files are analyzed as post-processing.
<b>Automobile's audio system</b>	Signal input available (FM, Bluetooth streaming, ...)	On-line detection and compensation of delay of used input channel in dynamic mode
	Autonomous playback from CD/storage in head unit	Static mode provides detection of response data. Unique IDs ensure analysis of correct playback channel
<b>Ceiling speakers in airports, malls, trains, ...</b>	Mobile recording	Static mode provides post-processing of wave files which are recorded with hand-held recording

device at the receiving position.  
Unique IDs ensure analysis of  
correct response data

## SUMMARY

In measurement situations with unknown, varying or unpredictable delay between playback and capture processes, the analysis window needs to be aligned to the response data. This synchronization process is required to be robust, fast and universal. *Klippel's* SYN module provides an advanced synchronization technique with simple setup integrated smoothly in the QC system. Even for complex test situations an exact synchronization, optimal placement of analysis windows and accurate measurements can be ensured.