Delay Measurements

XL2 optimizes reinforced sound

Modern conference rooms offer audio systems for best speech intelligibility of all persons in the room. Usually main- and supporting speakers are build up. Some of you may have experienced sound coming from the side speakers but actually the person is talking in the front, so the visual and acoustical perception are not matching. Achieving such a directional sound perception from the front is rather tricky.

The XL2 Audio and Acoustic Analyzer offers a helpful solution to achieve best performance easily. This application note describes practical examples.

Basic terms

Velocity of Propagation or velocity factor is a parameter that characterizes the speed at which an electrical or radio signal passes through a medium. Electrical audio signals are "traveling" in cables with the speed of the light, app. 300'000 km/second.

The speed of sound is a term used to describe the speed of sound waves passing usually through air. The speed varies with the medium employed as well as with the properties of the medium, especially temperature. At sea level, at a temperature of 15 °C (59 °F) and under normal atmospheric conditions, the speed of sound is 340 meter/second (m/s).

Why do we have delays?

As an example, in a large room with a length of 100m the electrical signal passes the distance in the cable in approximately 0.003micro seconds whilst the traveling time in the air is about 290ms. The difference is the so called "propagation delay". In reality we may consider the traveling time in cables as being negligible.

The challenge of sound re-enforcement

The human voice organ is not sufficient to supply the information to all seats in bigger auditoriums with an acceptable S/N ratio. As the speech intelligibility is getting reduced at lower S/N ratio and the sound energy is decreasing by 6dB at doubled distance, many conference rooms require sound reinforcement systems.

Unfortunately it is not as simple as putting down a few cables and have the speakers installed. Why?

The reinforcement speakers are much closer to the listeners in the rear and therefore provide the dominant portion of sound. Consequently, their perceived impression is that the human speaker is at the loudspeakers position. This divergence between natural sound source and reinforced sound is distracting and unnatural.

In addition, due to the propagation delay of the natural front wave, the speaker's voice is often perceived as an echo thus increasing the unpleasantness of the perception and decreasing intelligibility.

Here comes the Haas Effect into the picture, which helps to understand and solve these problems.

The Haas effect – Introduction

The Haas effect is based on a psycho acoustic masking effect:

- Localization of a sound source is based on the first wave front arriving at the listener's position.
- A second wave front arriving not later than 35ms after the first wave front supports at a higher sound pressure level supports the speech intelligibility.
- A second sound wave arriving later than 35ms is perceived as an echo.
- The second wave front can be up to 10dBASPL louder than the first wave front.

The binaural "location finder" allocates the sound origin based on the first wave reaching the ear.

The consequences are to ensure the original speaker's voice is reaching every listener before the second sound wave from the supporting speaker.

The support speakers have to be controlled to send their signals later than the front wave. For this reason we have to delay the support speakers.

The Haas effect – Basics of psycho acoustics

We assume the listener is within the direct sound field of the speakers (the direct sound is noticeably higher than the reflected reverberant sound), so the Haas effect can be utilized in practice as follows:

- Two sound waves from different directions arriving within less than 35ms are perceived as one sound wave. This is the time frame to set the delay for the supporting speakers: Minimum 10ms, maximum 35ms behind the first wave. As long as this criteria is met the reinforced energy is perceived to come from the front speaker and the aural and visual perception is in-line.
- The above is applicable even if the reinforced sound energy from the supporting speakers is up to 10dBASPL higher than the original first wave. This effect is particularly helpful as each support speaker can deliver more energy per position and thus reduces the overall number of support speakers required. A good practical rule says that the level of the re-enforcement speaker shall be set 6dB above the first wave front.

So the complete delay line alignment is a combination of delay and level measurements, which shall be explained step by step in the following samples.

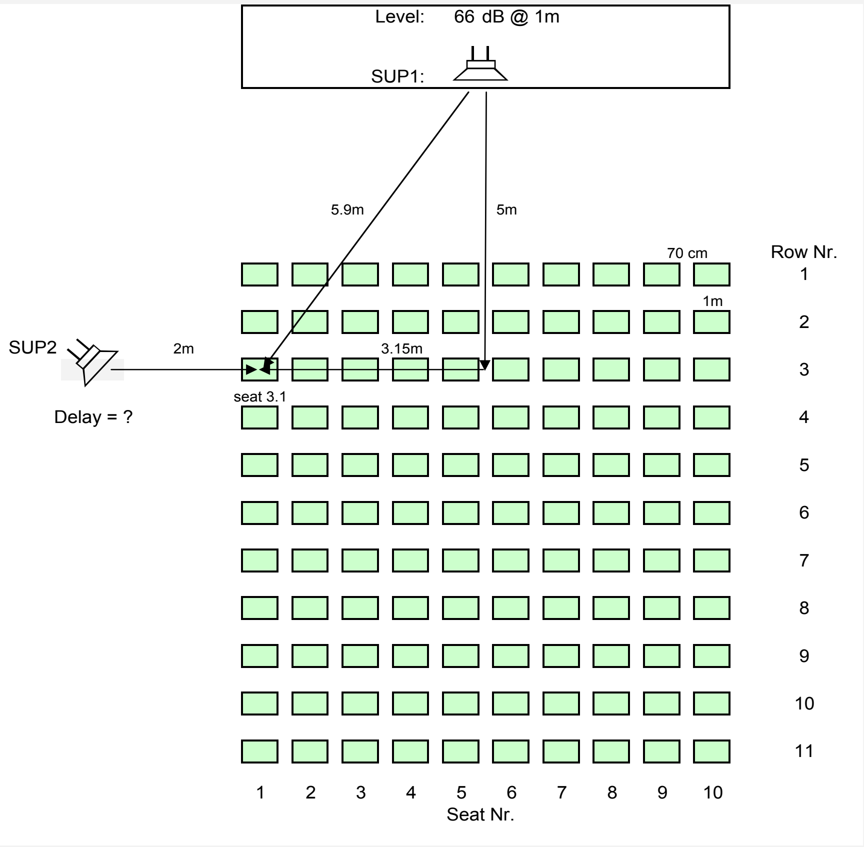
Sample 1

Situation:

See picture 1 for reference.

- In front of the auditorium is a speaker SUP1 (simulating a human person). The front speaker is set to 66dBA SPL @ 1m.
- At the left side is the support speaker SUP2. For simplicity the speaker has a radiation of 90° only.

What is the delay and sound pressure level of the support speaker SUP2 ?



Picture 1: Auditorium, sample 1

Measurement location:

Apparently row 3 is of interest as the mix of SUB1 and SUB2 is interfering. Seat 1 is the closest seat to the support speaker. Let's calculate therefore

- All relevant parameter of seat 3.1 (row 3, seat 1)
- The sound pressure level in the last row, seat 11.1

1. Calculations, Seat 3.1

Distance to SUB1 is 5.9m. The level of the front speaker decreases by 6dB per doubled distance.

- SPL of SUP1 at seat 3.1 = 66dB - 6*log(5.9m)/log2 =
  = 66dB - 15.4dB = 50.6dBA SPL

The acoustical delay at seat 3.1 from the front speaker SUP1 is calculated as follows:

- Acoustical Delay SUP1 = distance d / speed of sound =
  = 5.9m / 340m/s = 17ms

Seat 3.1 is in distance of 2.0m from the support speaker SUP2.

- SPL of SUP2 at seat 3.1 = 66dB - 6*log(2.0m)/log(2) =
  = 66dB - 6dB = 60.0dBA SPL
- Acoustical Delay SUP2 = distance d / speed of sound =
  = 2.0m / 340m/s = 6ms

Result:

The SUP1 sound arrives after 17ms at seat 3.1, which is 11ms later than the sound wave from the SUP2 speaker. So the person at seat 3.1 perceives the sound source from the left side but sees the actual person speaking in the front! The level of SUP2 is also 9.4dB louder than the front speaker and the localization is again underlined to come from left supporting speaker.

Improvements for seat 3.1:

Set the delay time of SUP2 to 11ms (= Delay SUP1 - SUP2) and both sound sources arrive at the same time at seat 3.1. To ensure the sound source is perceived to be from the front, we add an additional 5ms margin, so setting the delay of SUP2 to 16ms.

The level of SUP2 shall be maximum 10dB louder than the arriving level from SUP1. The default setting of SUP2 = 66dBA SPL @ 1m meets this requirements already.

Improvements for all seats:

The loudness localization effect is worst at the seats closest to the support speaker SUP2 and the delay setting is most sensitive to the most distant seats from the SUP2 Speaker.

To optimally set the support speaker SUP2 we have to ensure that for all seats the following applies:

- The sound wave from the support speaker SUP2 has to arrive after the sound from SUP1 in the front.
- The level of the SUP2 should exceed the level of the front speaker not more than 10dB.

The above calculations can now be executed for all seats, especially the critical seats.

2. Calculations, Seat 11.1, distance to SUB2 is 8.3m

- SPL of SUP2 at seat 11.1 = 66dB - 6*log(8.3m)/log(2) = = 66dB - 18dB = 48.0dBA SPL
- Acoustical Delay SUP1 = distance d / speed of sound = = 13.4m / 340m/s = 40ms
- Acoustical Delay SUP2 = distance d / speed of sound = = 8.2m / 340m/s = 24ms

The sound wave of SUP2 shall arrive about 5ms later then SUP1 for best directional recognition. SUP2 shall be set with a delay of 21ms.

Supporting tools:

The XL2 Audio and Acoustic Analyzer replaces the pocket calculator and measurement tape in this application. It even takes changes of sound speed into account based on changing environment temperatures. In a matter of seconds the above results are measured using the handheld XL2 Audio and Acoustic Analyzer.

An Excel sheet for download is available from the NTi Audio website www.nti-audio.com/XL2. This offers an abstraction of the room and conducts all the calculations of distances from centre speaker, distance from support speaker, the level at the listener's position and the corresponding delays for every seat. It also applies the Haas law for each seats and probes whether the conditions are met or not. In case the conditions are not met the seat color is changing and the status changes.

You may now say: "Oh, yeah! This is easy and can be done also for larger rooms", but are you sure?

Sample 2 - Using a bigger audience

We setup and optimize the sound re-enforcement system in an auditorium for 300 people with dimensions per attached sketch on page 8.

The lecturer speaks from the front, using two re-enforcement loudspeakers SUP1, positioned to left and right. Furthermore we find two sound re-enforcement loudspeakers at 6m (=SUP2) and 14m (=SUP3) distance on each side of the room.

For the simplicity, we consider no reflections from the rear walls.

Here are the questions we need to answer to optimally set the systems for best sound quality:

- How loud do we set the front speakers SUP1?
- Do we need a delay setting for SUP1 or do we feed the undelayed signal to them?
- What are the best levels and delays for the SUP2 and SUP3 speakers?

Basically we have two possibilities. Both approaches will be discussed and compared:

- Intuitive approach
- Measurement approach

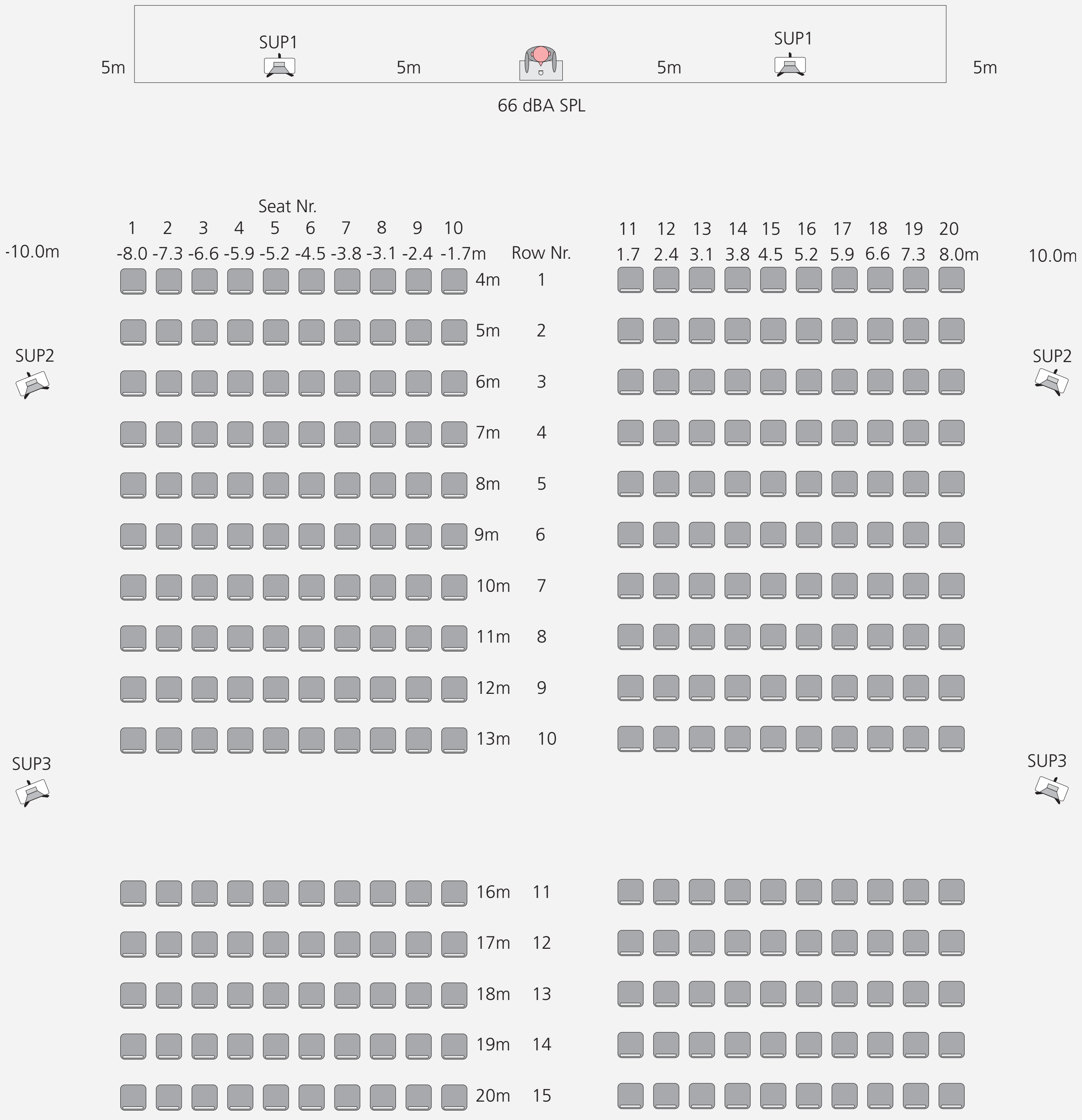
A) Intuitive approach

- The setup of the front speakers SUP1 will be without delays.
- The delay for the support speakers SUP2 in row 3 is set to 6m/340m/s = 18ms.
- The third speaker line SUP3 behind row 10 is set to 14/340m/s = 41ms.
- The level of the reinforcement speakers shall not be louder than the main speaker SUP1.

Results:

	Delay	Level @ 1m
SUP1	0	66dBA SPL
SUP2	18ms	66dBA SPL
SUP3	41ms	66dBA SPL

The result and effects will be discussed at the end.



Picture 2: Auditorium, sample 2

B) Measurement approach

First let's think which are the mostly affected seats? – The combination of seats being closest to the loudspeakers and those being most distant from the lecturer.

The outmost seats in row 1 and 20 are the most critical for the delays as their positions relative to the speakers are the extremes.

First we pick seat 1.1 (row1, seat 1, ❶ distance to lecturer is 9m). We use a centre speaker to simulate the lecturer.

Procedure:

- Activate a centre speaker to simulate the lecturer.
- Apply the delay measurement chirp, either from the NTi Audio Test CD or the Minirator.
- Select the DELAY measurement function of the XL2. The display reads "SYNC to cable" saying that we have now to synchronize the XL2 with the test signal. A few seconds are required for synchronization.
- As soon as the bar graph turns black, the delay measurements may start. You can disconnect the cable and move around the room to measure the delay on-line.

Set SUP1 Speaker

Delay setting:

- Now position yourself with the XL2 at seat 1.1 (❶) and get the delay reading of the centre speaker. It is 26.3ms and the distance reading below is 8.9m.
- Switch the centre speaker off and SUP1 active only. The delay at seat 1.1 is 14.7ms -> the sound wave of SUP1 arrives 11.4ms earlier than the lecturer's voice at seat 1.1.

The minimum SUP1 delay shall be set to 12ms, recommended 17ms (including an additional 5ms safety margin) for best directional sound recognition at seat 1.1. This ensures that the speaker's voice is arriving before the sound of SUP1. In all the positions in the first row the distance to the speaker is shorter thus improving the early arriving of the speaker's voice.

Level setting:

The Haas effect describes the level of the support speakers can be up to 10dBA SPL louder than the original sound. We measure at seat 1.1 (❶).

- Activate the centre speaker, pink noise at 66dBA SPL @ 1m.
- Select the SPL/RTA mode at the XL2. Result = 47dB at seat 1.1.
- SUB1 can be up to 10dB louder than the center speaker. Switch of the centre speaker and activate SUP1 with a pink noise test signal. You may increase the level until you measure 57dBA SPL at seat 1.1. This level is equal to 70dBA SPL @ 1m.

Results:

	Delay	Level max.
SUP1	17ms	70dBA SPL @ 1m

Set SUP2 Speaker

Next we set the support speaker SUP2. The seats 3.1 (❷) or 3.20 are critical; the distance from seat 3.1 to the centre speaker is 10.0m.

Procedure:

- Activate the centre speaker with the delay chirp.
- Measure the acoustical delay at seat 3.1. Test result = 30ms. The sound wave of SUP2 speaker must arrive later than 30ms at seat 3.1.
- Switch the centre speaker off and activate SUP2 with the delay chirp.
- Measure the propagation time of SUP2 at seat 3.1. (distance = 2.0m). Test result = 6ms. So SUP2 must be delayed by 30ms – 6ms = 24ms to comply with the Haas law.
- The recommended setting is 24ms + 5ms (margin) = 29ms.
- The level is set according to the nearest SUP2 seats in row 3. Measure the level with centre speaker and SUP1 active using pink noise. The SUP2 speaker is deactivated. The level in seat 3.1 is 54dBA SPL.
- So we may adjust the level of the SUP2 speaker at seat 3.1 to maximum 64dBA SPL. We perceive the voice coming from the lecturer in the front. (Level of SUP2 = 70dBA SPL @ 1m)

Results:

	Delay	Level max.
SUP2	29ms	70dBA SPL @ 1m

Set SUP3 Speaker

The third step is to align the speakers SUP3 that serves the back rows 11 to 15. Critical for the delays are again the most outside seats in row 11 (❸).

Procedure:

- Activate the centre speaker with the delay measurement chirp.
- Measure the acoustical delay at seat 11.1, distance is 17.9m. Test result = 53ms. The sound wave of SUP3 speaker must not arrive before 53ms at this point.
- Switch the centre speaker off and activate SUP3 with the delay measurement chirp.
- Measure the non-delayed propagation time of SUP3 at seat 11.1, distance is 3m. Test result = 8ms. So SUP3 must be delayed minimum by (53ms – 8ms = 45ms) to comply with the Haas law. The recommended setting is 45ms + 5ms (margin) = 50ms.
- The level is set according to the nearest seats to the speaker in row 11. Measure the level with central speaker and SUP1 active but the SUP2 speaker deactivated. Test result = 47dBA SPL.
- So we may adjust the level of the SUP3 speaker at seat 11.1 up to maximum 57dBA SPL. We perceive the voice coming from the lecturer in the front. (-> Level of SUP3 = 66dBA SPL @ 1m).

Results:

	Delay	Level max.
SUP3	50ms	66dBA SPL @ 1m

You may also refer to the available EXCEL calculation sheet for the complete room.

It is available for download at: www.nti-audio.com/XL2.

Measured versus intuitive comparison

Now we have found all the answers to the above questions and compare the results of the intuitive and measured approach.

	Intuitive	Measured
SUP1 level	66 dB	70 dB
SUP1 Delay	0 ms	17 ms
SUP2 Level	66 dB	70 dB
SUP2 Delay	18 ms	28 ms
SUP3 Level	66 dB	66 dB
SUP3 Delay	41 ms	50 ms
# of seats in spec	60 (20%)	290 (97%)
# of seats out of spec	240 (80%)	10 (3%)

The intuitive setting ends up that almost all seats perceive the sound to come out of loudspeaker rather than from the lecturer itself, compromising the perceptions and concentration. Up to 80% of all seats are out of specs.

The measurement approach using the XL2 Audio and Acoustic Analyzer for actual delay- and level testing provides 100% of all seats with best directional perception and speech intelligibility.

Try your setting

Both listed samples are available in a MS Excel sheet. You can setup individual speaker levels and delays simulating live conditions. The actual status of every seat is directly displayed in the visualized room sheet.