

HOW TO...

Set up for a Live Sound Event



This document provides a practical guide on how to use NTi Audio instruments in a Live Sound environment.

What you need: **XL2 Kit for Front of House**

- XL2 Analyzer
- M4261 Measurement Microphone
- Minirator MR-PRO
- ASD Cable
- Cable Test Plug
- Mains Power Adapters
- Exel System Case



XL2 Kit for Front of House

What to Do:

There are four areas where the NTi Audio instruments come in very handy,

- Finding faults
- Optimizing the frequency response
- Identifying feedback frequencies
- Ensuring compliance with legal requirements

Plus, if you have the requirement, you can

- Set up Delay Speakers

How to Find Faults

Everything is set up and switched ON and you are ready for the sound check, but there is a hum on the PA, or no sound from one of the microphones or instruments.

NO SOUND

To isolate where the signal path is being broken, choose a source point in the signal path, for example the microphone, remove the XLR connector and plug the MR-PRO into the down-line section of the path (towards the mixer).



Confirm that the line is providing Phantom power to the microphone if appropriate (the DC value should be $48\text{ V} \pm 4\text{ V}$).

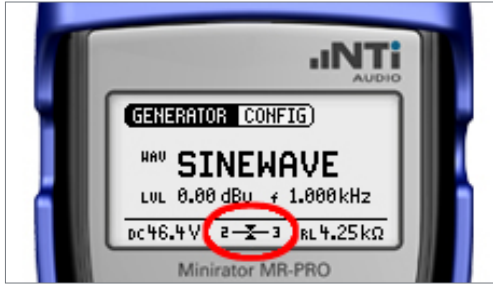
Select an appropriate sound, such as “FILE, MESSAGES, Adjust”, and listen for the signal through the PA.



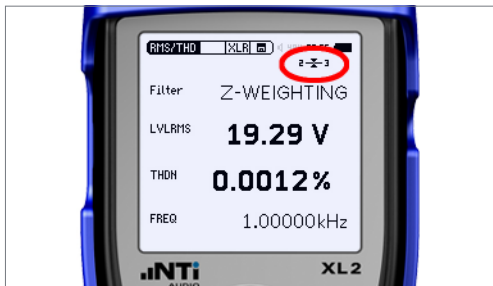
Interrupt the signal path at various points, plug the XL2 into the up-line section of the path (from the MR-PRO). Select “SLMeter”. Press the speaker button on the XL2 and listen for the signal from the MR-PRO through the XL2 internal speaker. To adjust the volume, hold the speaker button and turn the wheel.

HUM

To isolate which component is causing the hum, choose a point in the signal path, for example the stage breakout box, and remove the XLR connector.



Plug the MR-PRO into the down-line section of the path (towards the mixer). Select “SINEWAVE, 1 kHz” and check that the down-line signal is balanced, because a missing screen on the cable or an unbalanced line can often cause the hum.

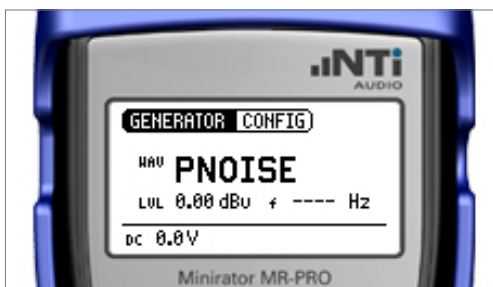


At the same point in the signal path, plug the XL2 into the up-line section of the path (from the mic or instrument). Select RMS/THD, create a sound at the source (ask the guitarist to play or gently tap the microphone) and check that the up-line signal is balanced. Press the speaker button on the XL2 and listen for the hum through the XL2 internal speaker.

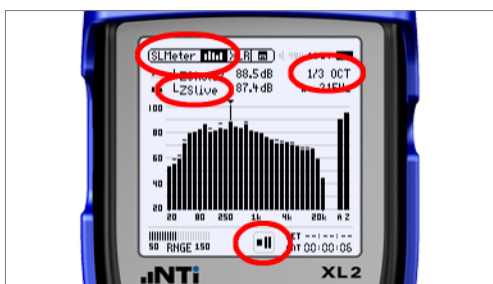
For tips on finding faults, see Appendix A

HOW TO OPTIMIZE FREQUENCY RESPONSE

Configure the system to deliver a similar response across the frequency spectrum for left and right (and center) of the PA.



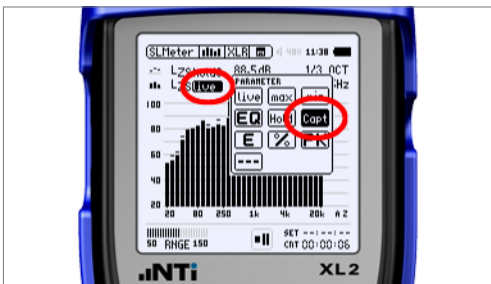
Play Pink Noise from the MR-PRO through the PA.



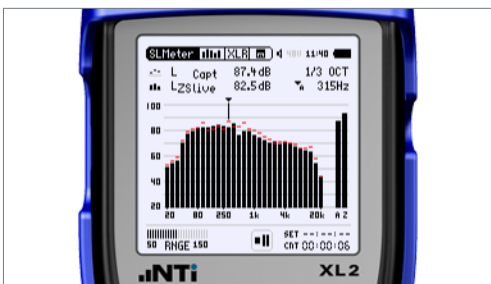
Select “SLMeter RTA” on the XL2, measuring LZSlive. Select 1/3 Octave resolution, as this gives you 31 bands of resolution, matching most traditional equalizers. Position the XL2 microphone at the nearest audience position in front of the left speakers. When you are satisfied that the frequency response of the left PA has no significant spikes or dips, press pause on the XL2 to freeze the spectrum reading from the left PA.



Capture the “LZSlive” spectrum as your reference spectrum.



Recall this Capture spectrum.



Position the XL2 microphone at the nearest audience position in front of the right (or center) speakers. Release the pause to measure LZSlive. Adjust the frequency response until the spectrum is similar to the reference spectrum of the left PA (the left PA Capture is shown here in red).



Play the dynamic speech and instrument sounds of the MR-PRO through the PA to get a feel for the whole sound.

Monitors

If you have frequency control over individual monitors, you can do the band (and yourself) a favour and configure each monitor to deliver a similar flat response across the frequency spectrum, using the method for optimizing the PA as described above.

HOW TO IDENTIFY FEEDBACK FREQUENCIES

To identify the frequencies that are causing feedback, choose the SLMeter RTA page (📊) of the XL2. Measure “LZF hold3”. This will hold the feedback frequency for a period of 3 seconds after it has occurred.

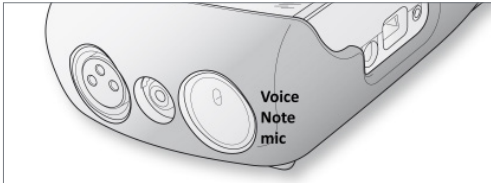
For tips on reducing feedback, see Appendix B.



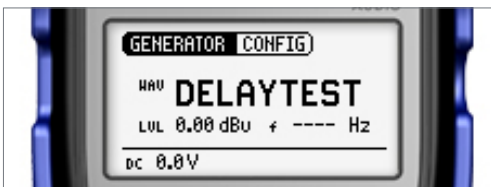
XL2 Analyzer with M4261 Measurement Microphone

How to Set up Delay Speakers

In order to determine how much delay should be added to the delay speakers, we need to measure the difference in time, between the front PA and the delay speakers, for the sound to reach the audience position closest to the delay speakers.



NOTE: Before the Delay test, disconnect the measurement microphone from the XL2 as the internal Voice Note microphone is used during this test.



Play the DELAYTEST signal from the MR-PRO connected directly to the XL2 in order to synchronize the two instruments. When synchronized, the XL2 shows that you are now ready to measure the delay with the word "Measure". You now have 5 minutes to measure the delay before another synchronization is required.



Play the Delay signal from the MR-PRO through the left delay speaker only. Stand at the audience position closest to the delay speaker and press the XL2 Delay STORE function. This sets your zero point.



Now play the Delay signal from the MR-PRO through the front left PA speaker only. Remain at the audience position closest to the delay speaker. Add 15 ms to the delay difference shown on the XL2 (81 ms + 15 ms = 96 ms in this example), and apply this delay value to the delay speakers. Assuming the speaker layout is symmetrical, apply the same value to the right delay speakers, or measure again on the right.

Why do we need delay speakers? - see Appendix C

Appendix A

Tips on finding faults

There are several actions you can take before reaching for your test instruments:

- Stay calm – blaming people or equipment will not solve the problem
- Check Stereo – is the problem on one side of the stereo image? – fault is probably post-mixer.
- Be methodical – follow each signal path, rather than checking components randomly
- Switch it out – consider changing multi-core channel, mixer channel with ones that are known to be working

NO SOUND

Check the obvious

- are all components switched on?
- is the channel muted on the mixer?
- are all cables in the signal path plugged in correctly?
- does a battery need replacing?
- is the input gain on the mixer too low?

HUM

Very often mains hum is caused by incorrect cabling of equipment

- if your earth reference voltage is different from one end of the signal path than the other – plug all components (mixer, instruments and PA) into a single electricity mains socket
- if you run an audio cable through a load of coiled mains cable
- if you use lighting dimmers on the same supply as your sound system supply

Earth hum can be very difficult to find, especially with very large PA systems. You can determine where the hum is coming from by unplugging all equipment from the inputs of your mixing desk, and then plugging in each piece of equipment one by one.

Appendix B

Tips on reducing feedback

Once your frequency response is optimized, you can identify potential feedback frequencies by ringing the system out. Push channels up individually, beyond the level at which you expect to operate during the show, until feedback is heard. Use the XL2 “SLMeter RTA” page to identify the feedback frequencies. Use a filter or an equalizer to reduce these frequencies.

Hint: Jingling car keys or clapping hands into the microphone can help trigger a feedback for testing.

There are several actions you can take to reduce the likelihood of feedback:

- Move the microphone, the monitor or the PA
- Use a more directional microphone
- Turn down the monitor volume
- Turn down microphone gain and ask the singer to hold the mic closer to their lips
- Look for reflective surfaces that might be bouncing the monitor or PA sound to a microphone and adjust as above
- Turn off microphones when not in use

Appendix C

Why do we need delay speakers?

If you can not cover the whole audience area with one speaker system, you can set up additional speakers on each side of the venue halfway into the hall. These speakers then cover the back half of the hall. All good. Problem is that the poor audience sitting just to the right (and left) of these additional speakers will see the performance in front of them but *hear* the performance from the speaker to the side of them. The result is that the sound experience for these people is not ideal – it is much more desirable to perceive the sound coming from the direction of the band. We can bring the perceived direction of sound back to the front PA while still maintaining enough level to cover the back half of the hall.

The trick is to add a delay to the additional speakers (thus called “delay speakers”) so that the sound from the front PA arrives at the audience closest to a delay speaker at least 15 ms *before* the sound from the delay speaker arrives. As the first sound arriving is then from the front PA, the perceived direction of the main sound is therefore also from the front PA, while the sound from the delay speaker is perceived as a reverb, resulting in a rich sound experience. Further, it is now possible to increase the sound pressure of the delay speaker up to 10 dB higher than the front PA without affecting the perception of sound direction.



Humans perceive direction of sound to be coming from the position from where the first sound arrives. If a sound from a second source (in this case the delay speakers) arrives within 5 ms after the first source, then this second source also has an influence on the perceived direction of the sound and the listener will perceive the sound to be coming from a position somewhere between the delay speakers and the front PA.

If a sound from the second source arrives between 5 ms and 35 ms after the first source, then this second source is perceived by the listener as a reverb and has no influence on the perceived direction of the sound – the sound is thus perceived to be coming from the direction of the front PA. If a sound from the second source arrives more than 35 ms after the first source, then this second source is perceived by the listener as an echo and the listener perceives two separate directions of sound – from the front PA and from the delay speakers. Giving the delay speakers between 5 ms and 35 ms (as a rule, set it to 15 ms) of delay allows you to bring more sound level to the back of the room without altering the perception of direction of sound. This somewhat remarkable concept is known as the “Haas Effect”.