ISCE

The Institute of Sound and Communications Engineers

Engineering Note 25.1

Sound System Test, Alignment, and Certification – Part 2

Order of works

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Sound out loudspeakers

This is a check that all loudspeakers on every circuit are actually functioning. This test proves that the cabling and termination is correct and that the loudspeaker itself is functioning.

Note: In environments where the access to the loudspeaker is difficult, loudspeaker functionality should be checked prior to installation in order to alleviate embarrassing re-visiting of the loudspeaker installation.

Feed a sound source (not music, but tone or pink noise) and check that the correct loudspeakers are working on a circuit by circuit basis at a relatively low sound pressure level. This can be done by feeding the source directly into the amplifiers before connecting them to the routing and distribution part of the system. It can even be done using a single amplifier which is temporarily connected to the loudspeaker circuit.

Routing and zoning

Most systems of any size have multiple zones which can be selected to receive different sources, possibly simultaneously. Proving that the loudspeaker circuits serve the desired zone is in effect done when the circuits are sounded out as described above. However it is still necessary to prove that all available sources can be routed to all relevant zones.

Feed a sound source at low level (not music, but tone or pink noise) into an input and route it to each zone on a zone by zone basis. Repeat for all other inputs.

In the case of a system which is capable of routing multiple sources to multiple diverse zones simultaneously, the systems engineer should use different sources at low level (mixture of tones and music and/or messages) and route them to appropriate zones. The sound engineer should then visit each acoustic space to check the validity of the routing.

It is very unusual that routing systems are wrong, but especially in these days of programmable signal processors, there can easily be slip-ups which go undetected until tested in reality.

Maximum level Part 1

This is an alignment necessary if there is a primary signal control, such as a digital signal processor, which is responsible for protecting the loudspeakers under load by means such as limiting and/or compression.

Note: this also applies for systems where the protection is provided within the amplifier hardware and, obviously, the spirit of the wording, rather than the order of the wording should be applied.

With any gain control on/in the amplifiers set to full power, increase the gain on the DSP until the correct SPL is achieved in that area on those circuits (bearing in mind that the circuits are likely to be interleaved) according to the electro-acoustic specification. This may require the interleaved circuits to be set for relative level prior to setting the maximum level. Set the compression and limiting to be capable of producing that level only as a maximum.

Relative level

This is an alignment necessary when there are multiple circuits in the same acoustic space. This may be simply a matter of interleaved circuits in a voice alarm system but in addition it could be because of time delay with respect to a focal point source.

With a feed level considerably below maximum but at a level which is probably 6 dB above ambient as a minimum, send a pink noise signal to a pair of adjacent areas. Measure the SPL in those areas and check that they are the same or adjust until they are. Progress on to the next adjacent area and repeat the measurement, having muted the previous area, until all acoustically linked areas have been covered.

Equalisation

This may be done in the same visit as relative level.

This alignment is to adjust the response of the sound system to its acoustic environment with respect to the loudspeakers' ability to reproduce at all relevant frequencies.

Check the response using the real time analyser function on the meter and make any equalisation adjustments if necessary.

Note: for larger systems where the control of equalisation is made from a central location, the use of radios is often necessary.

Acoustically-similar spaces should be anticipated and the system engineer should be able to make the equalisation adjustments in advance to save time.

Output delay

Where areas are always served by a single source and they are in the same acoustic space as another set of loudspeakers, the closer loudspeaker circuit requires a delay at the output. This means that the sound from the distant loudspeaker arrives at the listener at the same time, or very shortly after, the sound from the nearer loudspeaker.

Note: If the two sounds arrive at very different times (differing by more than 35 ms) the sound will be unintelligible.

Delay is one of the most difficult adjustments to make and requires site quietness, considerable concentration and no interruptions.

There are meters which measure time delay. By comparing a reference signal, (which needs to be a 'chirp' or 'drum click') taken directly from the amplifier to the meter, with a signal from an individual loudspeaker, picked up by a microphone in the meter, the meter can then show the time delay. At a large site, the signal from the amplifier can be connected via a radio link.

The delay times measured on the meter for all loudspeakers audible at a given point can be reported back to the system engineer, who can then introduce the appropriate amount of delay for each individual circuit, so that the time differences between the sounds arriving at that given point are less than 40 ms. The time delay may optionally be shown on the meter (or adjusted on the signal processor), expressed in milliseconds, or feet or metres, where 1 metre approximately corresponds to 3 ms in time.

If this is not possible, the approximate distance should be measured off drawings and converted into a delay time (330 metres per second, or 1 foot per millisecond). A drum track should then be used as a source signal and by listening and changing the delay time back and forth the correct delay should be established. **(N.B. This is a challenging task and not everyone is humanly capable of doing this, so don't be disappointed if you are not one of these.)**

Note: Delay is usually a feature of a larger system. From experience, it is particularly important when setting delay using radios that the system engineer does not confirm the setting immediately after making the adjustment. This is because the sound engineer in the acoustic space will be concentrating hard on whether the sound is now correct, and the radio will destroy that concentration.

Source delay

Where areas are served by more than one source, the delay needs to be placed at the front end of the system and each source set for each location.

The same test methods apply as for output delay.

Maximum level Part 2

It is quite possible that the adjustments above have altered the overall maximum level available to be below that required in the specification. For this reason, spot checks should be made and adjustments made accordingly.