Novel Methods for Acoustic Indoor Measurements and Applications in Aero-Engine Test Cells

P.Böhning [1], H. Siller [2], K. Holland[3], F. Arnold [1], A. Kempton [4], U. Michel[2]

[1] Rolls-Royce Deutschland, Dahlewitz, Germany
[2] DLR Engine Acoustics, Berlin, Germany
[3] ISVR, Southampton, U.K.
[4] Rolls-Royce, Derby, U.K.

ABSTRACT

The research institutes DLR in Germany and ISVR in Great Britain are developing together with Rolls-Royce, one of the major aero-engine manufacturers, methods for acoustic indoor measurements. These methods will allow acoustic tests to be performed in semi-reverberant aero-engine test cells. Currently, static noise tests are performed on open-air test beds in order to ensure free-field condition. However, open-air tests are prone to delays due to unfavorable whether conditions. Also, expensive noise protection installations might be required if the test bed is located close to communities.

The presentation first introduces two novel methods developed by DLR and ISVR. Their effectiveness will be demonstrated using the results obtained in simulations and in experiments using loud speakers as noise sources. Both methods will be verified by comparing the radiated far-field noise of a Rolls-Royce BR715 engine measured under free field conditions and the far-field noise estimated from measurements taken in one of the standard development and production pass off test beds at Rolls-Royce Deutschland in Dahlewitz. This final test is going to be performed under the framework of the German research project LEXMOS.

1 INTRODUCTION

The German Aerospace Center (DLR) and the Institute of Sound and Vibration Research (ISVR) in the U.K. together with Rolls-Royce are developing methods for acoustic indoor measurements of aero-engines. These methods will allow freefield noise levels to be deduced from engine tests in semi-reverberant test cells.

Currently, static noise tests are performed on open-air test beds in order to ensure acoustic free-field conditions. Limited availability of suitable test beds and testing restrictions due to unfavourable weather conditions and environmental noise issues, however, introduce significant operational problems not present with indoor testing.

This paper describes two novel methods of deducing freefield noise levels from measurements taken in semi-reverberant test cells under development at DLR and ISVR. The effectiveness of the methods is demonstrated using the results obtained in simulations and in experiments with loudspeakers as noise sources. Both methods will later be verified by comparing the radiated far-field noise of a Rolls-Royce BR700 engine measured under free-field conditions with the far-field noise estimated from measurements taken in one of the standard development and production pass-off test beds at Rolls-Royce Deutschland in Dahlewitz.

The novel methods are being applied to the specific characteristics of the indoor test bed at Rolls-Royce Dahlewitz. However, a later application is planned for indoor test beds with higher reverberant times.

In this paper, first, the specific room acoustic properties of indoor test bed at Rolls-Royce Deutschland will be described. Secondly, the novel methods will be explained and experience gained from a preparation test will be discussed. Finally, the measurements previously conducted on an open-air test bed and currently planned on an indoor test bed to validate the methods are described.

Future publications will describe further experiments with Rolls-Royce Trent engines that have been conducted within the European research project SILENCE(R), allowing more indoor testing in fully reverberant environment, and also comparison between different outdoor source location methods applied by ISVR and DLR.

2 ROOM ACOUSTIC PROPERTIES OF THE TEST BED AT ROLLS-ROYCE DAHLEWITZ

Before the actual indoor engine test, acoustic measurements using loudspeakers as sound sources were performed in one of the engine test cells of Rolls-Royce Deutschland at Dahlewitz,. The purpose of these tests was to benchmark the acoustic properties of the room and to test different microphone set-ups developed by DLR.

2.1 Indoor test bed at Rolls-Royce in Dahlewitz

The test stand at Dahlewitz is a tunnel of 70 m length with a square cross-section of 8 by 8 m, see Fig. 1. The actual test cell is 25 m long. Upstream of the test cell an air intake silencer and a pressure droop screen are installed. Downstream of the test cell an exhaust augmenter of circular cross section heavily lined with sound absorption material is attached. The huge amount of sound absorption material minimises community noise and the reverberant time inside the test cell. The tunnel can be closed by roll doors upstream of the intake silencer and right downstream of the test cell.

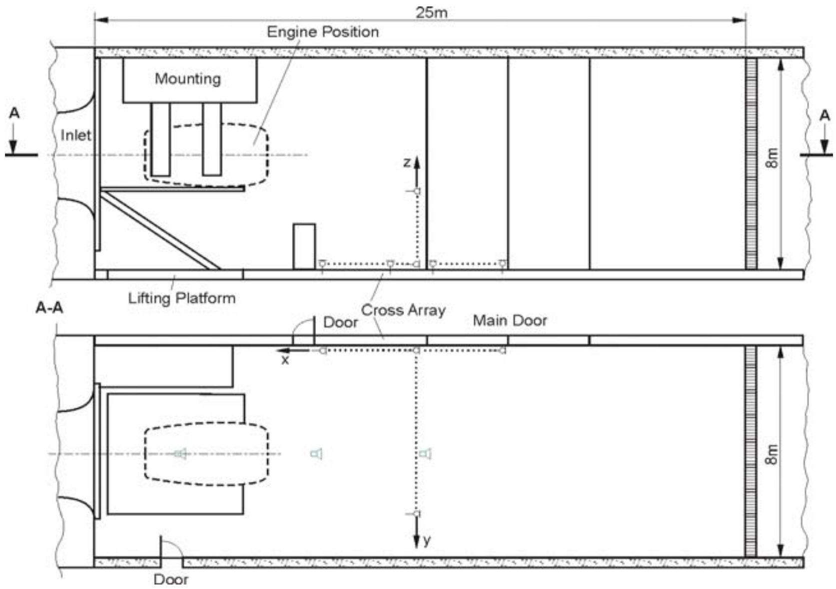

Fig. 1. Schematic of the engine test stand at Rolls-Royce Deutschland, Dahlewitz with DLR loud speaker positions and microphone arrays

The roof and one side wall are acoustically treated by a layer of mineral wool covered with perforated sheet metal. Strong reflections are expected from one wall and the floor. Sound scattering is caused by the engine itself, its mounting and additional internals, such as metal stairs.

2.2 Reverberation time and radius

The reverberation time T_{60} is a measure to describe the absorption of acoustic energy in an enclosure. It is the time during which the sound pressure level drops by 60 dB after a sound source was suddenly turned off. A strong absorption or a short reverberation time causes low background noise levels and increases the acoustic reverberant radius r_H [1]. It can be controlled by the amount of absorption material installed in a room.

The reverberant radius r_H defines the distance from a source where the pressure level of the direct sound and the diffuse noise field are equal. Equation (1) defines r_H for an omni-directional source and microphone [1]:

$$r_H = 0.057\sqrt{V / T_{60}} \tag{1}$$

where r_H in m is a function of the room volume V in m^3 and the reverberation time T_{60} in s.

The signal of a microphone positioned at a distance from the sound source smaller than the reverberant radius is dominated by the sound source. Microphones at larger distances, however, are dominated by the diffuse field or background noise. Here, the signal-to-noise ratio needs to be increased by a spatio-temporal averaging or cross-correlation of microphone signals at different positions.

Fig. 2. The engine test cell at Rolls-Royce, Dahlewitz with the omnidirectional sound source, individual microphones and the microphone arrays

An omni-directional sound source was used to determine the reverberation time in the room according to EN ISO 3382 and EN ISO 354. A noise signal was applied to the loudspeaker and suddenly turned off. From the 6 micro-phone signals distributed over the room, the reverberation time T_{60} was calculated. Fig. 3 shows the results for all one-third-octave bands between 100 Hz and 10 kHz for the two cases with open and closed roll doors. No significant difference was found for the two cases. The reverberation time for frequencies higher than 300 Hz is below 0.5 s, which is a surprisingly good result, considering that usual studio quality prescribes a maximum time of 0.2 s.

The corresponding reverberation radius was calculated with an estimated volume of the room V = 8 x 8 x 25 m and is plotted as a function of frequency in Fig. 4.

Assuming that an engine is the sound source and all microphones are positioned at the test bed walls, r > 4 m, all microphones signals are located in the diffuse field in the entire frequency range. It can be concluded that single microphone measurements are not suitable. Spatio-temporal averaging or cross-correlation of microphone signals at different positions is necessary.

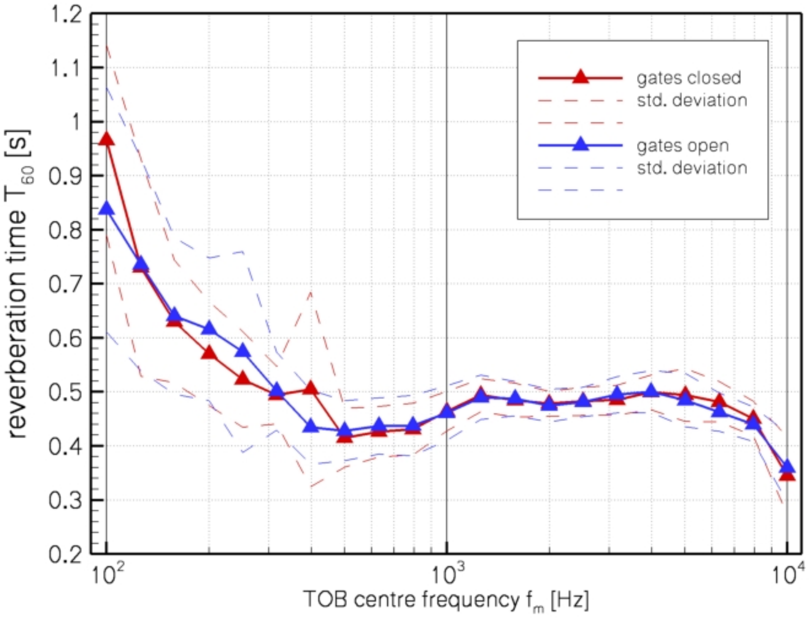

Fig. 3. Reverberation time in the test bed as function of the one-third-octave-band centre frequency

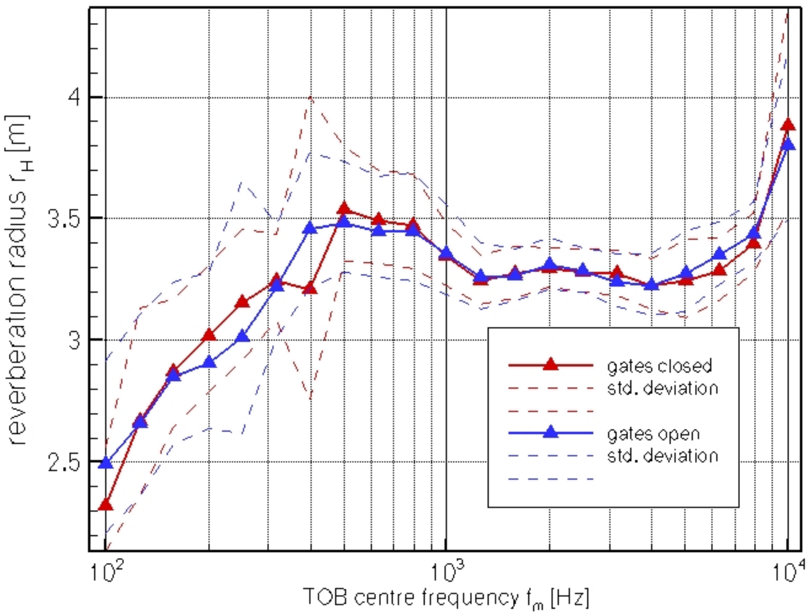

Fig. 4. Reberberation radius in the test bed as function of the one-third-octave-band centre frequency

2.3 Standing waves

Standing waves might cause a strong dependence of pressure magnitude on the microphone position and could lead to wrong results. However, if the number of room modes N present in a certain frequency band Δf is much higher than 1, pressure maxima and minima are averaged out and the sound field can be approximated by a diffuse field.

The number of modes present in a rectangular room within a one-third octave band f_c can be estimated by

$$
\begin{aligned}
&N \approx \left[4\pi V \left(\frac{f_m}{c}\right)^3 + \frac{\pi}{2} S \left(\frac{f_m}{c}\right)^2 + \frac{l}{8}\left(\frac{f_m}{c}\right) \right] \frac{\Delta f}{f_m} \\
&V = l_x l_y l_z \,;\, S = 2(l_x l_y + l_y l_z + l_x l_y) \\
&l = 4(l_x + l_y + l_z)
\end{aligned}
\tag{2}
$$

with V being the volume, S the surface and l the edge length of the room [2]. It seems to make sense to evaluate the number of modes each one-third octave band individually. Fig. 5 shows the estimated number of modes per one-third-octave band in the test cell. According to this plot no significant influence of standing waves is expected above $f_c > 100$Hz.

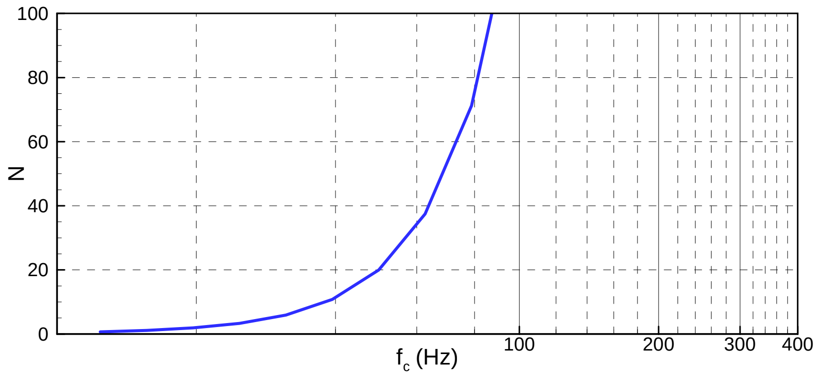

Fig. 5. N umber of acoustic modes per one-third-octave-band potentially present in the test cell

In order to experimentally validate this conclusion the cell was excited with a broad band noise by means of an omni-directional sound source located far away from the array at *(x,y,z)* = (6.55;4;2.7) [m]. The averaged sound pressure spectrum of all microphones is plotted in Fig. 6. Only levels at $f_c > 31.5$Hz are dominated by the excitation noise signal.

If standing waves were present, the pressure magnitude would be expected to show a strong dependence on the microphone position. The one-third-octave band levels are plotted in Fig. 7 as a function of the position of the microphones along the *x*-axis.

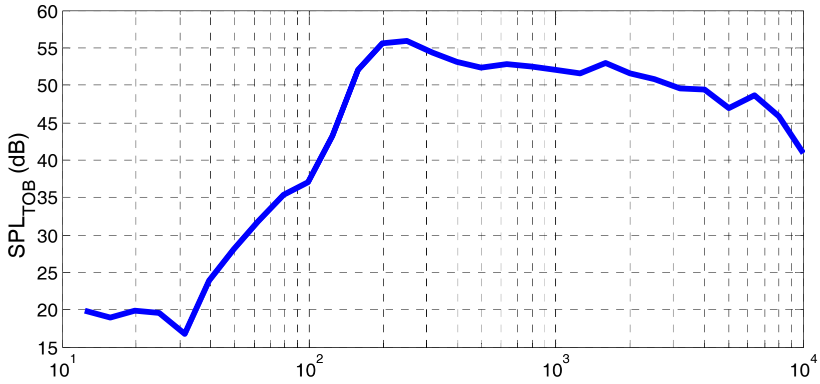

Fig. 6. One-third octave band spectrum of the signal used for the acoustic mode study (in arbitrary dB scaling) averaged over all microphones of the linear array along the x-axis

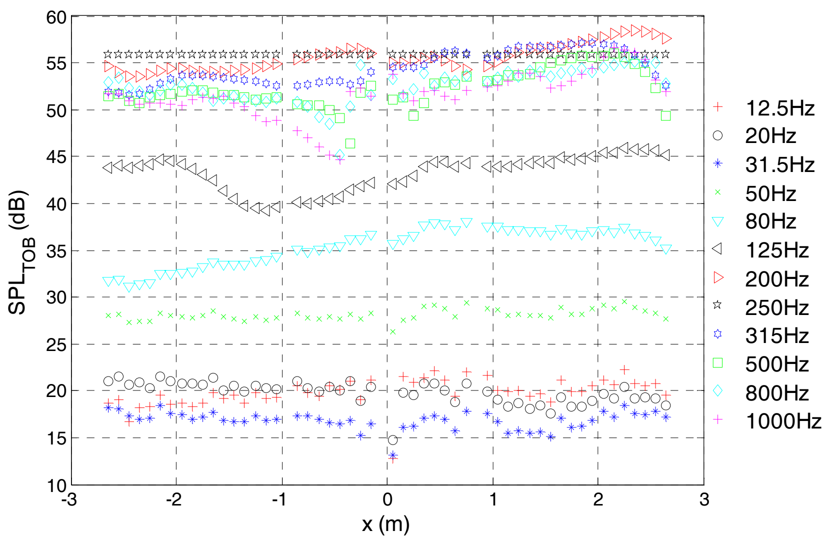

Fig. 7. One-third octave band level with normalisation at f=250Hz (arbitrary dB scale)

Only microphones placed in the edge between floor and wall (y=z=0m) are regarded. The signals are corrected with reference to the one-third octave band levels at f_c = 250Hz since no microphone channel calibrations were available (the original levels are plotted in Fig. 8).

The levels at frequencies $f_c \leq$ 50Hz vary only little. Between 80Hz $\leq f_c \leq$ 125Hz, however, a strong dependence on the microphone position is observed which can hardly be produced by standing waves. Between 200Hz $\leq f_c \leq$ 500Hz, again, only very small level variations are found which might be caused by slight variations in the distance between the microphones and the wall. The average distance was Δy = 5cm. At f_c = 1000Hz, the levels suddenly jump from 45dB to 52dB at about x = -0.4 m, where the frame of the test cell door is located. The sudden increased in the distance of the microphone to the wall causes the discontinuity.

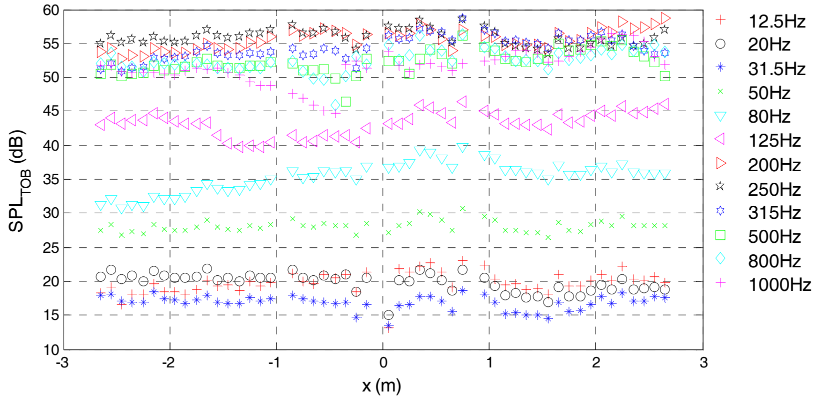

Fig. 8. One-third octave band levels without calibration as function of x (arbitrary dB scale)

Some of the variation can still not be explained satisfactorily and will be studied during the BR700 test in detail. If this dependence of the levels on the microphone position can not be resolved, it might be necessary to perform the source localisation with normalised microphone signals.

2.4 Acoustic near field and far field

The source to microphone distance is limited by the test bed dimensions. A microphone placed in the acoustic near-field, however, could suffer significantly from pressure fluctuations which do not propagate as sound. Assuming a far-field criterion of $r > \lambda/2$, where r is the distance from the source to the microphone and λ the sound wave-length, a microphone on the test bed wall ($r = 4$ m) can be expected to be in the far-field for frequencies $f > 42$ Hz. This frequency range would meet the requirements for engine noise measurements.

2.5 Air flow properties

The mean flow produced by the aero-engine inside the test cell is different to the mean flow on an open-air test bed. Therefore, the sound propagation of the radiated noise is different and needs to be taken into account when far-field noise or noise source localisation is calculated. The mean flow profile in the test cell was measured in a previous test at different power settings.

Boundary layer turbulence is expected to cause only a slightly higher level of uncorrelated noise at the microphones since flow velocity is low near the wall. It will be compensated by spatial and temporal averaging or cross correlation. This issue has already been discussed in [3].

The turbulence rate of the mean flow may have an minor impact on fan noise and sound propagation. It was measured for the BR700 but will be subject of a later investigation.

Studies on the impact of turbulence on the quality of source localisation were already performed for wind tunnel [4] and open-air facilities [5].

3 ACOUSTIC MEASUREMENTS IN INDOOR TEST-CELLS

The indoor measurement methods should allow the far-field noise characteristics of an aero-engine, such as spectral power density and directivity, under free-field condition to be estimated from the sound field observed in a test cell. However, measurements at indoor test beds are complicated by the acoustic properties of the room. The reflections of the sound waves at the walls make the sound field diffuse and also introduce ambiguities into sound source localisation measurements.

Both methods presented here firstly assess the sound source distribution and secondly estimate the far-field sound from this data. It should be mentioned that other methods avoiding the determination of the sound source distribution are imaginable. Unfortunately, the classical beamforming for sound source localisation does not necessarily work in reverberant rooms due to the coherent reflections. A similar problem occurs in acoustic measurements with microphone arrays in wind tunnels. Strong reflections on the wall cause wrong source distributions when data processing is performed using the classical beamformer for free-field conditions. Guidati et. al. [6] developed a modified beamformer, the Reflection Canceller, which takes the first reflections into account. Its precision is limited at higher frequencies by uncertainties about the correct positions and reflection factors of the mirror sources to be cancelled. Moreover, optimal position and reflection factor might be a function of frequency. Better results may be obtained by a method proposed by Sijtsma and Holthusen [7], which derives position and strength of the mirror source from a minimisation between measured data and a mirror source model.

DLR’s method uses a special microphone arrangement, the classical beamformer and the Reflection Canceller. ISVR’s method is based on the measured transfer functions between the position of the sources and the microphones. This approach eliminates the influence of the room on the sound field.

It should be mentioned that in a first step only the far-field characteristics of the entire aero-engine are of interest. These characteristics are reviewed when the acoustic impact of engine design changes need to be evaluated. The noise characteristics of single sound sources, such as fan or core noise, will be targeted in a later development.

3.1 Indoor array method of DLR

A microphone inside a room will receive the direct sound from a source as well as the reflections from the walls. Sound source localisation based on the classical beamforming which considers only the incident sound wave may fail when the reflections have significant levels. The Reflection Canceller also takes the reflections into account, which improves the localisation results when the reflections are compensated correctly. The reflections are modelled by mirror sources. Their positions depend on the constellation of source, microphone and wall. Their complex amplitudes depend on the spectral reflectivity of the wall. Uncertainties in these parameters affect the quality of the source localisation.

The new microphone arrangement proposed by DLR consists of placing the microphones along the inner edge of the room between the floor and the adjacent wall. Thus, the microphone is only subject to direct reflections from the opposite wall and the roof. Fig. 9 shows how a microphone placed on the floor away from the side wall will see at least three wall reflections in addition to the source itself. However, when the microphone is placed at the edge between the floor and the side wall, only reflections from the opposite wall and roof can reach the microphone. The pressure sensed at a wall is two times and in the edge four times larger than under free-field conditions.

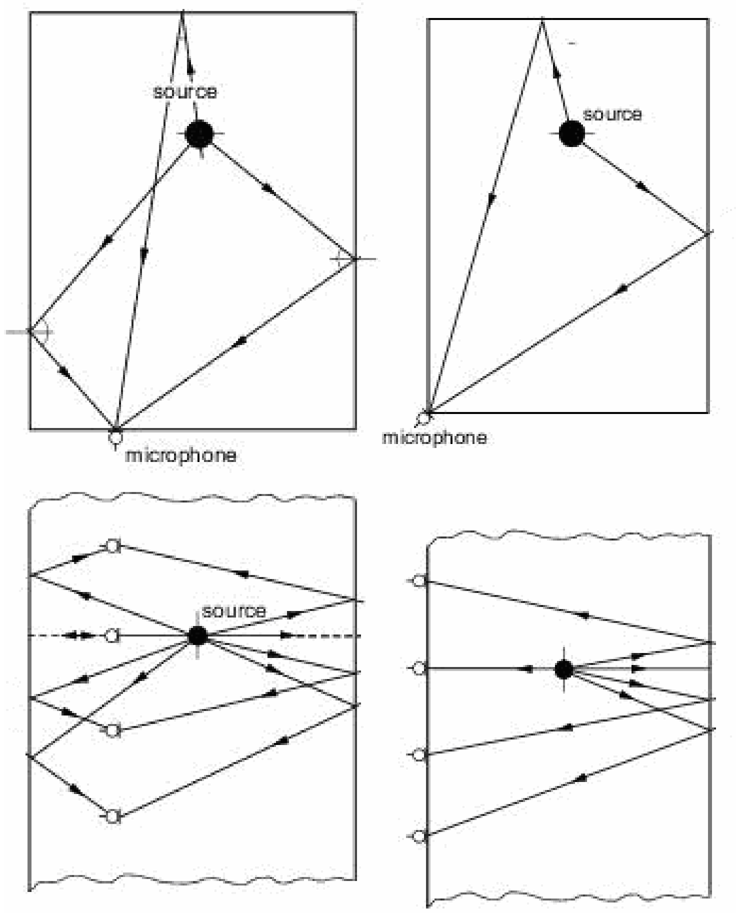

Fig. 9. Wall reflections sensed by a microphone on the floor and in the edge; Top: side view, Bottom: surface view

If the two remaining surfaces are acoustically treated, like they are in the stand at Dahlewitz, then the contribution of the reflections becomes very small. Fig. 10 shows the auto-correlations of the signal emitted by the omni-directional sound source with microphone signals measured at two positions in the room. One microphone was fixed at the inner edge between the floor and the side wall, the other was placed on the floor 0.72 m away from the wall. The data for the wall microphone shows the effect of the first refection of the sound waves on the wall. The sound pressure level of the reflection is only about 6 dB below that from the direct path. The microphone at the inner edge, however, does not show any direct reflections. The main reason being that the adjacent wall and floor cannot reflect sound to the microphone and the opposite wall and roof are covered with acoustic liners.

The absorber panels are not sufficiently effective at low frequencies. In the low-ferquency range, the sound source distribution on the engine axis will be calculated by the Reflection Canceller. At higher frequencies the classical beamformer be will applied. The signal-to-noise ratio is improved by a large number of microphones and by spatio-temporal averaging.

To validate this approach, three linear arrays consisting of a total number of 135 microphones were laid out in the test cell along the x, y, and z-axis (see Fig. 2). A loudspeaker was used as sound source at different positions in the test cell. The sound pressure level on the loudspeaker axis at r=1m was measured allowing the true and the estimated source strength to be compared.

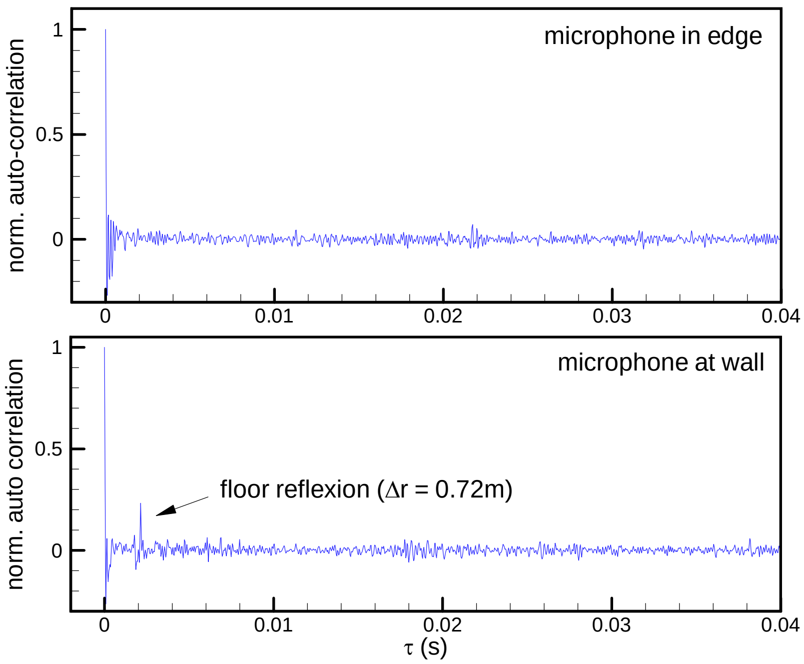

Fig. 10. The auto-correlation of a wall-mounted microphone (bottom) indicates early reflections, while the microphone at the inner edge (bottom) does not

First promising results from experiments with a calibrated loudspeaker and a linear array in the engine test-cell were obtained. The loudspeaker was placed 4.2 m from the wall with the array at an angle of 31° relative to the centre of the array. The array consisted of 54 microphones that were evenly spaced with a separation distance of 100 mm. In Fig. 11, the beamforming result is plotted against a simulation based on the array geometry, assuming a monopole sound source at the loudspeaker position under acoustic free-field conditions. The position as well as the amplitude has been determined quite accurately for the loudspeaker signal at 1 kHz. The far-field characteristics can be simply calculated by applying the free-field Green function [8].

The directivity of the source can be estimated focusing on the source at different angles by using a moving sub-array. A comparison with the directivity of the loudspeaker already measured in an anechoic chamber will be performed later.

For the real engine test, the favourable conditions in the test-cell might not be good enough in order to reproduce the results of the open-air free-field test. Then, extensions to the applied beamforming technique, such as proposed in [6], will be introduced.

Some experiences gained in the preparation test are also worth to be mentioned. 1) All microphones lay in the diffuse field of the room. Single microphone measurements are therefore insufficient to characterize the sound field of an aero-engine. 2) The quality of source localisation processed with the classical beamformer is good up to 2 kHz. So far, the lower limit has not been tested. It can be concluded that the reflection factor of the acoustical treated wall and ceiling is sufficiently small for the purpose of localisation. 3) The quality of source localisation decreases dramatically above 2 kHz. It was found that the gap between the microphones and the wall varied between 1 … 10 cm due to irregularities on the wall such as tubes, door frames. Therefore, it was proposed to use an artificial edge with pre-installed microphones consisting of two wooden boards combined in an angle of slightly more than 90°.

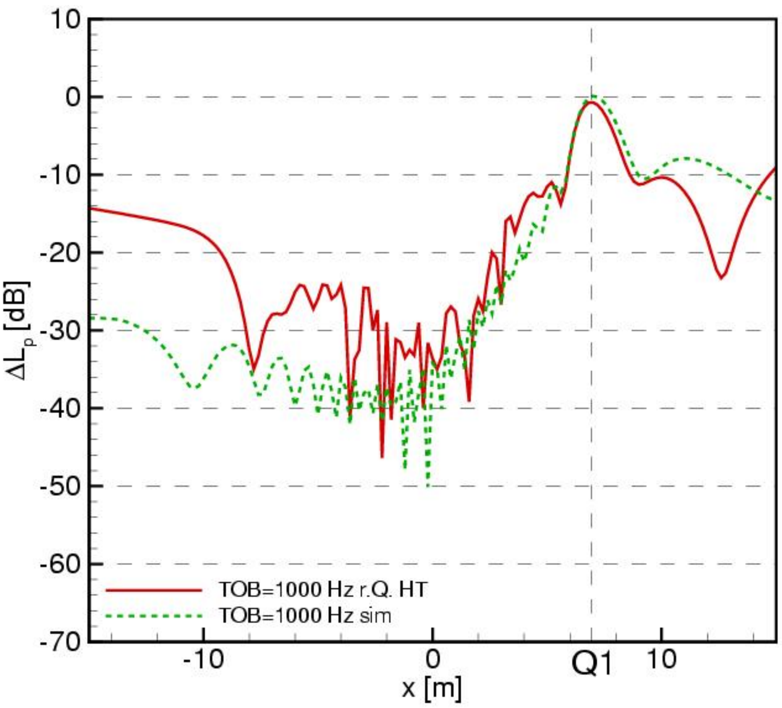

Fig. 11. Beamforming results for a 1kHz loudspeaker signal (solid line) with a simulated source (dotted line) under free-field conditions

3.2 Indoor array method of ISVR

Many engine test cells do not have acoustically-treated walls or ceilings, and are therefore very reverberant spaces having reverberation times of many seconds. For indoor acoustic measurements to be effective in these test cells, some form of de-reverberation is required. The reflection canceller mentioned earlier in this section can only deal with one or at best a small number of wall reflections and is therefore not particularly useful in highly reverberant spaces. The method proposed by ISVR for use in highly-reverberant spaces requires detailed knowledge of the Green functions which link the pressure at a microphone in the array and the source strength of a single, elemental source. The Green functions are defined thus:

$$p_m = G_{mn} q_n \tag{3}$$

where p_m is the pressure at microphone m due to source n having a strength q_n.

Under free-field conditions, one may estimate the strength of an omni-directional source by simply measuring the pressure anywhere at a known distance from the source. This is possible because the free-field Green function linking the source strength to the pressure is known. Indeed, classical beamforming under free-field conditions relies on estimation of the inverse of the free-field Green functions from a knowledge of the relative positions of the sources and microphones alone. In general, and in reverberant spaces in particular, the Green functions are not known and therefore need to be measured. For simple source distributions, the Green functions are best measured directly by substituting a source of known strength for each of the unknown sources in turn and measuring the pressures at each of the microphones for each source position. If the source positions are not known, the measurements are carried out at a grid of points to allow for beamformer scanning. For distributed sources the source region is divided into a number of source elements and the Green functions measured at the centre of each element. Alternatively, it may be more convenient to measure the Green functions reciprocally. This invokes the principle of acoustic reciprocity which states that the Green function linking the pressure at a receiver due to the output of a source in independent of an interchange of the positions of the source and receiver. In this case the known source is moved to each microphone position and the pressure measured at each of the positions of the unknown sources. The choice of directly- or reciprocally-measured Green functions is one of convenience.

In order to estimate the sound radiated by a distributed source, the elements into which the source region is divided have to be smaller than one half of a wavelength at the frequency of interest if spatial aliasing, and hence errors in the radiated field, are to be avoided. For large source regions, such as the inlet plane of an aero engine, this criterion can result in a very large number of source elements, particularly if sound radiation at high frequencies is of interest. Also, it is generally-accepted wisdom that in order to fully characterize a source region, the number of microphones required must equal or exceed the number of source elements. Satisfying both of these criteria could therefore require the use of very large arrays of microphones. In an earlier publication [10], a technique was described that exploits any known correlation structure within a source region to reduce the required number of microphones. This technique shows much potential in simulations and will be tested by ISVR on the sound radiated from the inlet of a Rolls-Royce BR700 engine installed in a test cell at Rolls-Royce Deutschland in Dahlewitz.

The inlet of the engine will be divided into 84 elements; the full source strength matrix of which will be determined from the output from an array of just 20 microphones arranged along one wall of the test cell. In order to test the de-reverberation method described above, the array will be mounted on the lined wall instead of the relatively reflection-free edge position of the DLR array.

As the actual sources of the sound radiated by the inlet are within the engine, these sources, which, of course, operate in the presence of flow, are replaced with a distribution of equivalent sources over the blocked inlet plane which operate without flow; the effects of flow on the Green functions is contained within the equivalent source distribution. This then allows the use of a set of Green functions measured without flow (measuring the actual with-flow Green functions would be very difficult). The task then is to use the outputs of the microphones along with a full set of measured no-flow Green functions, to determine the strengths of the equivalent sources which, under no-flow conditions, gives the same radiated sound field as the real sources in the presence of flow. This equivalent source distribution may then be combined with a set of no-flow, free-field Green functions to yield estimates of the free-field sound radiation from the engine inlet. It should be noted that this technique is expected to work for the broadband part of the radiated field only, as the source correlation assumptions are not expected to apply to tonal sources.

In addition to the inlet array, a second array will be installed by ISVR to investigate the sound radiated by the nozzles at the rear of the engine. The array will be arranged as a two-dimensional spiral mounted on the sidewall of the test cell alongside the rear of the engine. The output from this array will be used to determine the source strength distribution by scanning a beamformer over the nozzle area, and the total sound power output of the engine using space-averaged mean-squared pressure.

4 EXPERIMENTAL VALIDATION OF NEW ACOUSTIC MEASUREMENT TECHNIQUES

The novel acoustic methods for indoor measurements developed by DLR and ISVR will be verified by comparing the radiated far-field noise of a BR700 engine measured under free-field condition and the far-field noise estimated from measurements taken in one of the standard development and production pass-off test beds at Rolls-Royce Deutschland in Dahlewitz.

The reference measurement conducted on an open-air test bed of Rolls-Royce U.K. and the indoor test performed on an indoor test of Rolls-Royce Deutschland at Dahlewitz are detailed below.

4.1 Acoustic Measurement on open-air test bed as reference

The reference measurement was taken on the open-air test bed 11 of Rolls-Royce in Hucknall, U.K., in September 1999, [9]. Fig. 12 shows the mounted engine in flight configuration, i.e. with flight intake and Thrust Reverser Unit (TRU) installed. A Turbulence

Control Screen (TCS) was installed in front of the engine in order to control the inlet turbulence.

Fig. 12. Rolls-Royce BR700 aero-engine mounted on an open-air test bed in Hucknall

Three acoustically different engine configurations, which are listed in Table 1, were tested. They differ only in the acoustic lining applied in the intake and on the exhaust cone which, however, causes significant differences in the far-field sound field. Two noise records of 30 sec were taken for each of 19 power setting between low idle and high power after stabilising. The engine power setting is given as corrected low pressure shaft speed NL_c, equation (4), where NL is the mechanical shaft speed, T_{20} the fan face total temperature and T_{ref}=288.15K the reference temperature.

$$NL_C = NL\sqrt{T_{20} / T_{ref}} \tag{4}$$

The sound field radiated by the engine was measured by means of 31 far-field microphones positioned on an arc between 10° and 160° in intervals of 5° as shown in 0. The arc of radius R = 45.7 m (150ft) was centered on the engine's Center of Gravity (CG). In order to avoid ground interference distortion, ground plane microphones installed at d = 12.7 mm (0.5'') above the ground were used. Windscreens were deployed to all microphones to minimse wind effects.

Temperature and humidity were measured to be in accordance with ICAO Annex 16 and 14CFR 36.

The noise data analysis included averaged narrow band spectra for broad band noise and levels of engine order tones as well as one-third octave band spectra and directivities. The frequency range is f = 0 … 12 kHz and the bandwith of the narrow band analysis is Δf = 32 Hz. For the validation of the indoor measurement techniques, in a first step only differences between different configurations will be compared. In a second, step also the absolute noise levels will be validated.

Table 1. Test configurations of the BR700 engine

Build	Intake	Exhaust cone
997	Lined	Liner 1
998	Lined	Liner 2
1001	hard wall	Liner 2

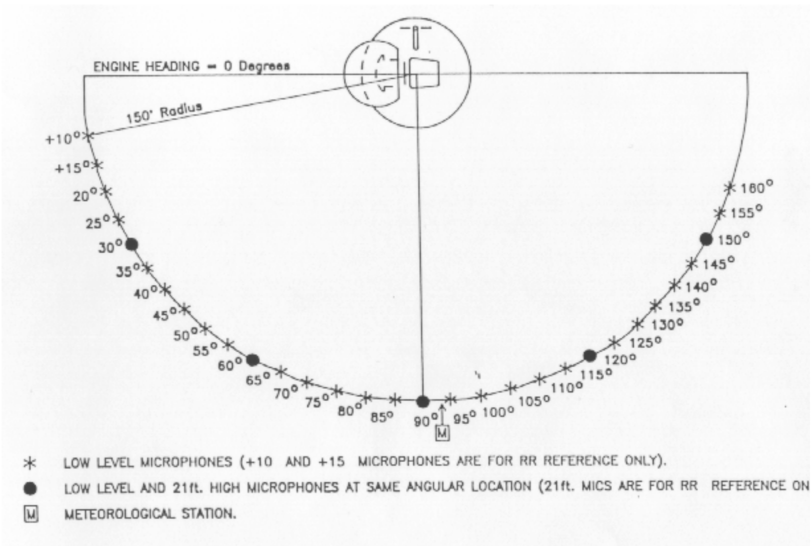

Fig. 13. Distribution of the far-field microphones used for the BR700 engine test on open-air test at Hucknall

5 ACOUSTIC MEASUREMENT IN AN INDOOR TEST BED

The sound field in the indoor test bed produced by the BR700 engine configured as in the open-air test were measured in September 2006. According to their methods, both research institutes, DLR and ISVR, deployed their own optimal microphone arrangements. In contrast with the open-air test, the possible microphone arrangements are limited by the test cell dimensions. Additional constraints were imposed by the door openings, the stairs and other internally mounted devices. Fig. 14 shows the microphone arrangements used for the indoor noise test at RRD Dahlewitz.

In the following some differences between the open-air and the indoor test bed, which might additionally affect the sound field, are listed: 1) Instead of a Turbulence Control Screen (TCS) controlling the inlet turbulence, a debris guard is installed in indoor test beds in order to avoid pieces being swallowed up by the engine. Due to the robust construction of the guard, it does not only change the inlet turbulence but also the sound propagation. Therefore, the debris guard was removed during the noise tests. 2) The mean flow field in the test differs significantly from the conditions on the open-air test bed and affects the propagation of sound. This, however, can be easily corrected by applying methods that are already in use for wind tunnel measurements.

Similar the open-air test, two noise records of 30 sec each were acquired for 10 power settings between low idle and high power after stabilising.

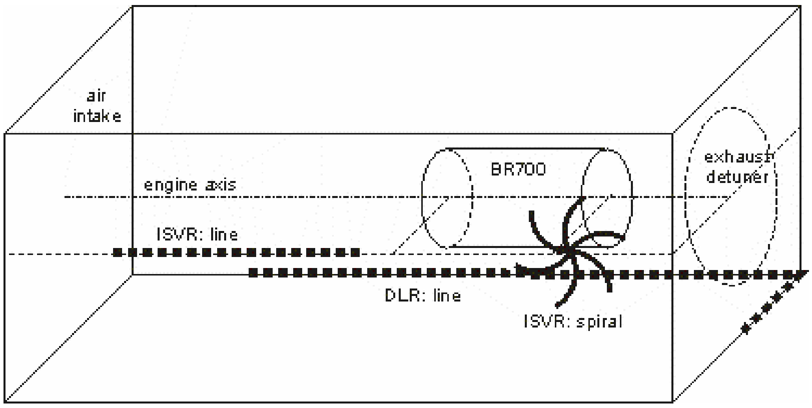

Fig. 14. Experimental set-up and microphone arrangement for the BR700 indoor test at Dahlewitz

5.1.1 Experimental set-up and method of DLR

DLR adapted their line-array method for sound source localisation on aero-engines [9] to measurements in a closed test-cell, where the microphones are now aligned along the inner edge between the floor and the wall to avoid the strongest reflections. Difficulties regarding reflections are only expected for the microphones in the rear corner. In order to ensure a high signal-to-noise ratio, a large number of microphones was used. A line array consisting of 72 equally spaced microphones was installed in order to measure the engine inlet, a second line array with 44 microphones was used to measure the exhaust region and extended around the corner at the downstream end of the test cell.

5.1.2 Experimental set-up and method of ISVR

ISVR will apply a spiral array and a line array both installed at the test bed wall at a height corresponding to the engine centre line. This wall is acoustically treated with sound absorption panels. The spiral array will be focused on the exhaust nozzle and the jet mixing zone right downstream of the exit plane. The line array will be used to estimate the radiation of the inlet.

The spiral array consists of 80 microphones pre-installed on a wooden board of size 2.5x2.5m. Its centre will be about 1m downstream of the nozzle exit plane. The line array consists of 30 equally spaced microphones attached to the wall at axial positions between 0 to 5m upstream of the intake plane.

A full set of 1680 no-flow Green functions will be measured by attaching 28 miniature microphones in a cross formation to a wooden board taped over the engine inlet. An omni-directional loudspeaker source will be moved to the microphone array positions and the outputs from the 28 microphones recorded for each of the 20 positions. The measurements will be repeated after rotation of the board through 30 degrees and then 60 degrees to simulate 12 radial lines of 7 source elements each.

6 CONCLUSIONS

Two novel advanced measurement techniques for indoor aero-engine noise tests have been developed by DLR and ISVR and evaluated with numerical simulations and tests with simple sources. Some useful practical experience has been gained during a preparation test on the indoor test bed of Rolls-Royce Deutschland at Dahlewitz. Measurements have been taken on an open-air test bed of Rolls-Royce in Hucknall, U.K. providing the far-field noise characteristics of a BR700 aero engine. These measurements will be compared with the free-field noise levels deduced from noise data to be taken in an indoor test bed, in order to validate the performance of the novel methods.

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