Proceedings of the Institute of Acoustics

BT's ACOUSTICS LABORATORY: PAST AND PRESENT.

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1. INTRODUCTION

The author has worked in transmission systems modelling and Electro-acoustics in BT (formerly the GPO) since 1967. Few people outside of the company realise that there has been a considerable commitment to acoustics since at least 1928, (that is the earliest reference I can find). The laboratory has undergone many metamorphoses since those early days and before it undergoes any more I felt that it would be useful to reflect upon and capture some of the engineering achievements. Hopefully it will also open a friendly window into what must seem to many to be a closed community. A more complete historical account is published in the recent IOA Bulletin. This article seeks to introduce the reader to the range of acoustical engineering carried out in the laboratory which is aimed specifically at improving speech intelligibility.

2. THE EARLY DAYS

The first long distance line in America was opened on the 7th of February, 1893 by AT&T. From that time onward, it became clear that there were numerous important patents and improvements pertaining to telegraphy and telephony drifting across the Atlantic. The engineering research station of the post office was already established at Dollis Hill in London N.W.2. and noted these patents with great interest. A laboratory was established to test out the usefulness or otherwise of the patents. It was soon realised that to objectively quantify a telephone system, a range of tools would have to be designed and built in-house. This had the added advantage that home grown improvements could be quantified just as easily. And so the acoustics laboratory was born, with a mission to continuously improve the performance of telephone instruments and to develop the art of measurement.

In 1919, Webster introduced the concept of acoustical impedance with reference to the strength of sound sources as applied to horns. In 1928 the principles of the measurement of acoustic impedance were developed further in the laboratory by the then senior engineer, Willie West. In 1932, Willie went on to write his classic book "Acoustical Engineering: The theory of sound and its applications to telephone and architectural engineering and to acoustical measurements and research". The book contains the distillation of many ideas which are commonplace today. For instance, an early measure of the efficacy of a telephone conversation is the the concept of intelligibility and is defined as "the ratio, or percentage, of the number of ideas received correctly to the number sent; and it is the reduction of intelligibility due to the transmitting system alone that is the desirable criterion of performance of the system". In the late 20's the staff of the acoustics lab drew heavily on Rayleigh's "Theory of Sound" and the Bell Systems Technical Journals and of course performed research of their own, Suffice to say that by the end of 1932 the lab had early versions of all the essential equipment for telephony measurements and transducer development. This list is not exhaustive but includes; Artificial ears, a clockwork level recorder, Speech level measuring equipment, probe microphones, microphone calibration equipment and an audiometer; all designed and built in the laboratory.

3. THE PRESENT DAY

BT's Acoustics laboratory now includes the Dynamic Analysis team, housed in purpose build facilities at Martlesham Heath near Ipswich in Suffolk. The dynamic analysis laboratory is based around a Ling Dynamics shaker system driven by a 24 KW water cooled amplifier housed in its own room. A separate control room houses the control equipment and provides a measure of sound insulation. Blinds are fitted to the windows to enable resonant searches to be carried out with a stroboscope. Transportational and destructive testing is carried out on a wide range of components from underwater cable joints to racks of exchange equipment to product packaging. We also provide building vibration consultation to the company with the overriding mission of maintaining and improving network integrity.

The acoustics lab consists of two anechoic rooms, one large, one small which work down to 60 Hz and 150 Hz respectively and a reverberation room. There are three associated measurement laboratories and a workshop. The facility is built into the side of a man made hill for temperature and vibration stability. Each anechoic room is mounted on replaceable metalastic supports and enclosed inside a further concrete structure to achieve noise levels as low as -15 dBA.

The reverberation room conforms to the ISO standard and is mainly used for generating diffuse sound fields - an essential requirement for measuring the performance of telephone handsets under noisy conditions. The room is further fitted out with a large perspex rotating paddle to remove standing waves and an equalised four channel noise source for the diffuse sound field.

One of the features of the large anechoic chamber is a flooring system which can be adjusted to the required size to support large objects such as a KX100 telephone kiosk. This type of kiosk has been experimentally modified to improve the reduction of externally generated noise by means of added attenuation and absorption. A reduction of some 12 to 14 dB has been achieved which is subjectively equivalent to more than halving the loudness of the environmental noise.

If necessary, the flooring can be removed altogether for maximum performance. This is certainly required for polar responses, such as the response of an experimental table top conferencing system developed in 1977. This system used a figure of eight ribbon microphone in the centre of the table and a cardioid microphone at the opposite end to the loudspeaker. To avoid a null in the polar response due to interaction between the two microphones, the outputs are separated by a phase shift of 90 degrees, analogue in this case.

At present the performance of the large anechoic room is slightly compromised by the addition of eight studio monitors for the replay of periphonic, ambisonic recordings. The recordings are mainly of background noise such as may be found at railway stations and other noisy public areas. The system is calibrated to unity gain so that the replay sound pressure is the same as it was when recorded. The system has been used to select optimum sending and receiving levels for mobile telephones by means of a subjective experiment. By replaying the same piece of background noise to the subject panel the variability of the results can be dramatically improved thus ensuring a low residual error.

The reflections from the loudspeakers have been quantified by using an artificial mouth as a repeatable wideband source and a 1/2" measuring microphone. Each is fixed to the end of an aluminium bar for dimensional stability. By measuring and storing the transfer function, f(a) under true anechoic conditions, the equipment can be relocated or reflecting elements introduced and a further transfer function measured, f(a+b) where b is the energy reflected from another object. By manipulation, the term f(b/a) can be determined which represents the reflected energy normalised by the transfer function of the transducers used. The amplitude frequency response gives the frequency and contribution of the reflection while the slope of the phase response gives the distance to the object. This equipment has been used successfully in a listening room to detect energy stored and released from vases and bottles etc. A useful feature is that at a given frequency the measurement indicates the whereabouts of the most active reflecting object. Once this is removed the next prominent reflecting objects at that frequency can be found.

The ambisonic system is based around a Calrec sound field microphone and an Audio & Design Digi-4 four channel digital recorder. The Digi-4 interleaves the bit streams from two Sony pcm 701 encoders and stores the signal onto a U-matic video recorder. The B format decoder was designed in house and feeds 8 JBL 4410 monitor loudspeakers via 150 watt amplifiers. High power is required because of the very low room gain in the anechoic conditions. The system has proved to be extremely reliable and the sound reproduction is second to none.

The recording sessions took place mainly around the London area and included sites where customers had complained that telephone conversations were difficult due to the high background noise. One such site is a K6 kiosk on the flightpath near Heathrow. The local residents had complained of high vibration levels. This was, in fact, the least of their problems.

The equipment which goes into a telephone kiosk has to be extremely robust as it will be in a hostile environment throughout its lifetime. The design of a payphone handset has to take many more design features into account than just the acoustical properties.

The new payphone handset is a case in point. This handset is the first to be designed by the acoustics lab specifically for payphone use. Features which are not immediately obvious include the use of a moving coil receiver. This gives a smooth frequency response combined with a good transient performance. The light weight of the diaphragm ensures that the mechanical shock resistance is high. An electret microphone combines robustness with low price. Both transducers are shielded from the environment with moisture barriers. The short scoop design is intended to lead the customer into positioning the handset correctly as it is difficult to position the scoop under the chin. The scoop also helps to create a pressure zone in front of the mouth to improve the sensitivity to speech as the high acoustic impedance of the electret can only ever sample the sound pressure.

Determining how people hold and use handsets is an interesting example of design by function through form. By which I mean that there is a high degree of interactivity between the handset and the user. An experiment has been carried out to determine, in a non intrusive way, exactly how people used handsets under different conditions and handset types. In order to measure the handset position a stereovision imaging system was constructed whereby once calibrated, if the same point can be seen in two image planes then the point can be determined in 3D space to the nearest millimetre. The entrance to the subjects ear canal is referenced to dots on the subjects face and the centre of the lips. Similarly, the major axes of the handset which pass through the plane of the earcap are referenced to light emitting diodes on the top of the handset. By this means it is possible to determine the position of all the major axes and dimensions, even though they are not visible when the handset is in use. Furthermore, the data sets can be manipulated so that they correspond to an erect head, aligned to a given axis and with the ear canal entrance as the datum. At this point it is then possible to average the data into the condition sets.

It has been noted that handsets with a low acoustic impedance receiver such as the Beocom are held in a different way to their traditional counterparts. When the handset is moved away from the ear, this acts as a volume control with little loss of the lower frequencies. These handsets are difficult to measure objectively using an ear simulator. This is because the airpath created between the handset and the pinna has to be included and is a highly variable quantity within subjects. The acoustics laboratory has a wide range of recognised and experimental ear simulators, each with their own limitations as to use. Work is presently in hand to develop a universal ear simulator which can be used for a wide variety of transducers, including 'Walkman' type headsets increasingly used in multimedia applications.

The IEC 711 ear simulator found in the newly specified Head and Torso Simulator (ITU-T P58) is probably the best available at present, but even this shows limitations if the impulse response is measured. By impulse response I mean the way that a pulse of acoustic energy entering the ear canal is treated. Intuitively in the real ear energy passes along the ear canal through the eardrum and some is reflected by the cochlear back through the eardrum. In the case of the IEC 711 simulator, the impedance simulation components are in front of the 1/2 inch pressure microphone. Energy stored and released in these components then applies a pressure to the microphone diaphragm in the wrong direction. There is a good case for developing a new ear simulator which has an open backed microphone diaphragm with the impedance elements further downstream. This is shown to improve the input impulse response and hence the simulation.

In order to measure acoustical impedance the laboratory has developed its own equipment known as AIMS (Acoustical impedance measuring System) This uses a high impedance source and a probe microphone to measure the sound pressure in front of the source. By storing the transfer response between the source drive and the probe output with a known and then an unknown impedance fitted it is possible to calculate the unknown impedance. Initially the known impedance was produced by a volume which yields a fixed compliance. The accuracy of the system has been significantly enhanced by treating the volume as a truncated open circuit transmission line. The line is always operated well below its first resonance but the input impedance calculation now yields a small but significant resistive component greatly improving the accuracy.

In conclusion I hope that the reader or listener has gained an insight into the nature and detail of a selection of the work carried out by BT's Acoustics Laboratory and where it fits into the drive for continuous improvements in reproduced sound. It is my personal view that with the continuing fall in the cost of bandwidth and improvements in perceptual coders, there are exciting possibilities ahead with conferees able to share the same virtual acoustic space.