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Evaluation and Performance Analyses of Wireless Audio Streaming Solutions for Musical Instrument Cable Replacement

BEng Electrical & Electronic Engineering Final Year Project Report

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Abstract

While wireless communication technology has been widely capitalised on in the consumer market for voice and data transmission, high fidelity audio streaming is a relatively newly-enabled and under developed application. The domain of wireless audio signal transmission for musical instruments for live monitoring purposes in particular places especially high demands on the communications system in terms of audio quality, robustness and latency.

The purpose of the project at the subject of this report was to investigate and evaluate the technologies currently employed by commercial instrument cable replacement solutions, conducting performance analyses on representative systems to characterise the effect of the communications technologies on delivered audio. The project also covers an investigation into the suitability of Bluetooth technology for this unique wireless audio application, which would enable exciting opportunities for interconnectivity with devices such as tablet computers and smartphones.

Developing and utilising experimentation instrumentation using the LabVIEW programming language in tandem with the National Instruments ELVIS measurement platform, a range of industry recognised audio system performance metrics were computed. The technologies investigated were found to have very different impacts on the nature of audio output from each representative system.

Project Specification

Title: Evaluation and Performance Analyses of Wireless Audio Streaming Solutions for Musical Instrument Cable Replacement

Supervisor: Professor W. G. Scanlon

Moderator: Dr M. van Walstijn

Areas: Audio, Wireless Communications, LabVIEW

Specification:
Several commercial systems exist for wireless transmission of musical instrument signals in live performance scenarios. These systems allow performers such as electric guitar players to freely move around the stage and interact with other artists and their audience, all without encumbrance from trailing leads. Two main types of system exist, those based on analogue transmission technology and those based on more modern digital technology. The purpose of this project is to perform objective assessments on the performance of commercially-available analogue and digital based wireless audio systems, particularly in the context of live scenarios where even low levels of latency may become important for both the artist and the others in an ensemble.
The investigation also covers the suitability of Bluetooth technology for the wireless instrument cable replacement application, considering issues such as the impact of the use of advanced codecs for audio compression for transmission across limited bandwidth digital links.

The project objectives are:

- Evaluate and justify the selection of candidate systems for testing.
- Thoroughly investigate Bluetooth modules from suppliers such as ST and CSR.
- Research metrics for assessing audio quality and select metrics which can be used to compare the systems.
- Produce instrumentation tools using NI LabVIEW/ELVIS board to compute the relevant quality metrics.
- Complete a performance analysis of the systems under a range of conditions to test the effects of environment and interference on audio quality.
- Consider and develop recommendations for overcoming any limitations in the systems evaluated and identify any opportunities for technical innovation in this unique audio application.

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Finally I would also like to thank Gareth Conway for his assistance and guidance in operation of the sweeping panels in the reverberation chamber at ECIT.

Declaration of Originality

I declare that this report is my original work except where stated.

Signed:

Date:

Table of Contents

1. Introduction

The purpose of the project documented by this report was to investigate the technologies employed under commercial wireless guitar cable replacement systems, conducting performance analyses on systems representative of their respective wireless technology in order to fully characterise the impact of each technology on the delivered audio signal. Completion of the project involved three main areas of work, categorised as research, testing (including instrumentation production), and analysis of experimental data.

Presented first is the findings of research conducted into the operation of wireless communications technologies, including traditional analogue transmission systems and newer digital technology, specifically Bluetooth. Also considered in detail are industry-standard audio system evaluation techniques and quantitative quality metrics, as well as methods for qualitative assessment of perceived audio quality.

Production of a methodology and associated instrumentation tools to test each system's performance is covered next. Experiments were conducted using the National Instruments Educational Laboratory Virtual Instrumentation Suite (ELVIS) prototyping and experimentation platform, controlled and automated where appropriate by custom-written Virtual Instrument programs in the LabVIEW graphical programming language. Demonstrating the design and operation of the hardware setup and software tools produced, and including descriptions of the environments in which testing was carried out, the full experimentation methodology is presented.

Test results are presented in graphical form for ease of visualisation. In most cases, measurements are made using a number of separate input fundamental frequencies, so that the performance of each system is characterised in detail across their entire frequency response range.

Finally, the results of data analysis are discussed alongside a critical evaluation of the experimentation methods employed throughout the project, considering the suitability of frequency-domain analysis and the hardware used for signal generation and data acquisition. As presented in the final conclusions section of the report, the project finds major differences in the performance of the systems tested, each of which displaying unique strengths and weaknesses over the others. However, it has been demonstrated that none are able to deliver in all three of the aforementioned requirements of quality, latency and robustness.

2. Project Objectives

Here the objectives of the project will be further clarified and expanded upon, in order that they may be fully understood and to facilitate the production of a Project Plan for the completion of the work.

- **Evaluate and justify the selection of candidate systems for testing.**

It is necessary to perform background research into the solutions currently employed on the market, in order to select appropriate systems for testing that are representative of their respective wireless communication technology (analogue or digital).

- **Thoroughly investigate Bluetooth modules from suppliers such as ST and CSR.**

To investigate the suitability of Bluetooth technology for the wireless instrument cable replacement application it will be necessary to acquire/develop a Bluetooth-based system. Therefore some investigation into Bluetooth technology and available hardware will be required so that a suitable system can be evaluated.

- **Research metrics for assessing audio quality and select metrics which can be used to compare the systems.**

It will be necessary to conduct some research into current techniques for assessing the quality of audio and audio system performance, including any applicable standards which may exist in this field. This will facilitate the selection of metrics allowing for meaningful comparison of the wireless systems.

- **Produce instrumentation tools using NI LabVIEW/ELVIS board to compute the relevant quality metrics.**

The design and production of software tools for to perform the relevant metric calculations shall serve to make the testing process more efficient by automating certain procedures and enhancing the repeatability of the tests.

- **Complete a performance analysis of the systems under a range of conditions to test the effects of environment and interference on audio quality.**

It is only by testing the systems under various conditions that their robustness can be determined, which will shed further light on the suitability of each technology for the specific application.

- **Consider and develop recommendations for overcoming any limitations in the systems evaluated and identify any opportunities for technical innovation in this unique audio application.**

Involving more research as well as the application of experiential knowledge, recommendations for improvement of the relevant technologies shall be considered.

3. Project Schedule

3.1 Gantt Chart

The project has been broken down into a series of general tasks to be carried out in order to complete the Project. A Gantt Chart has been produced to order and allocate sufficient timescales. This chart is shown in Figure 3.1.

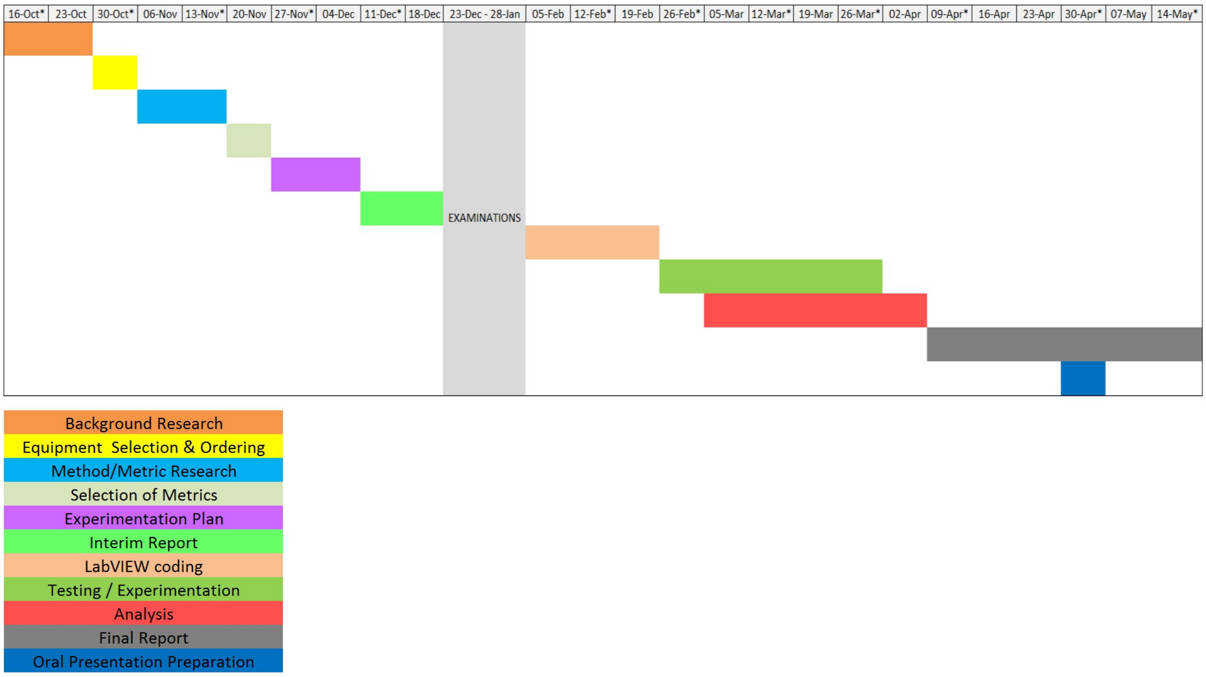

Figure 3.1 – Project Gantt Chart and Key

[NOTE: A larger version of the chart is included as Appendix 1.]

3.2 Task Breakdown

Following is a summary of the various tasks to be completed:-

1. Background Research

The first stage of the project will be to perform some market research to see the kind of technologies employed in existing commercial wireless instrument cable replacement systems.

2. Equipment Selection & Ordering

Applying knowledge gained from the market research, appropriate equipment can be selected and ordered for the completion of the project.

3. Method/Metric Research

Since the objective of the project is to assess the performance of the systems regarding Audio Quality, it will be necessary to perform some research onto methods used in industry for the objective assessment of audio quality.

4. Selection of Metrics

Selecting those audio quality assessment methods that apply to the project.

5. Experimentation Plan

The production of an Experimentation Plan shall outline the procedures to be employed in evaluating the performance of the various systems.

6. Interim Report

Writing of the project Interim Report.

7. LabVIEW Coding

Production of LabVIEW virtual instruments for experimentation.

8. Testing/Experimentation

Performing the experiments and tests as outlined in the Experimentation Plan.

9. Analysis

Performing analysis of the results of the experimentation, including creating LabVIEW tools to process the data.

10. Final Report

Production of the final Project Report.

11. Oral Presentation Preparation

Preparation for the project Oral Presentation.

4. Background Research

4.1 Initial Research

Electric instruments such as electric guitars and basses produce electrical signals by the principal of electromagnetic induction, whereby the vibrations of the instrument's metallic strings cause voltages to be induced in the conductive coils of the magnetic pickups. The induced signal is an A.C. (Alternating Current) wave, with frequency equivalent to the frequency of vibration of the strings. These low-level signals must then be amplified and converted back into sound energy by the same principal in reverse, where the amplified electrical signal is passed through a conductive coil in order to produce alternating-polarity magnetic fields, which act to attract and repel a permanent magnet, thereby reproducing vibrations at the same frequency as the strings of the musical instrument.

Traditionally, the low-level signals at the output of an instrument's pickups are transmitted to amplifiers directly via conductive coaxial cables. However, there are a number of problems with the use of such cables. For example, the signal can be distorted by induction of unwanted frequencies by the 50Hz mains electricity 'hum', although this can be protected against by using shielded cables. A more serious issue is that of unwanted Low-Pass Filtering, whereby high frequency components are lost due to capacitive properties of the coaxial cable. Since the total capacitance of a coaxial cable is proportional to its length, very long cables which facilitate free movement can have a significant detrimental effect on signal quality.

Wireless instrument cable replacement systems eliminate most of the problems associated with using physical cables, but introduce some of their own, which will be investigated in this project. All such systems comprise two main parts: a transmitter and a receiver. The transmitter is responsible for taking the signal at the instrument's output and generating a radio signal to be picked up by the receiver. The receiver then converts the radio signal back into the same electrical signal present at the output of the instrument, inputting that signal to the amplifier.

4.2 Analogue Systems

Analogue wireless communications systems operate by transmitting radio waves at a certain frequency which have been modulated (altered in some way) by the signal which contains information. In the case of wireless instrument cable replacement systems, the modulating signal will be that which is produced by the instrument.

The vast majority of analogue wireless audio systems use Frequency Modulation, whereby the frequency of the carrier is increased or decreased proportionally to the amplitude of the modulating signal (the instrument signal). The resultant modulated signal can have any frequency within a continuous range as defined by the Peak Frequency Deviation - the maximum difference in frequency from the central carrier frequency, measured in Hertz.

At the receiver, the instrument signal can then be recovered by demodulating the received RF signal. The instrument signal is fed to the output of the receiver to be input to an amplifier.

4.3 Digital Systems

Digital Systems operate by transmitting a digital representation of the instrument's signal. Using an Analogue to Digital Converter, the analogue signal generated by the instrument is regularly sampled, i.e. the instantaneous amplitude is recorded at regular intervals, where each sample is quantized or 'rounded' to the nearest of a number of discrete levels, digitizing the signal. Under Pulse Code Modulation (PCM), each of these discrete levels is encoded as a value in binary format such that the signal is transformed into a digital data stream.

The level of accuracy to which the input signal can be reproduced at the receiver is defined by the sampling rate and the bit-depth. The more bits used to encode each sample of the signal, the greater the resolution of the digital representation. This is because using more bits allows for a greater number of digitally-representable signal levels within the input signal range, and so the quantized samples will be closer to the actual signal amplitude.

According to the Nyquist-Shannon Theorem, the minimum sampling frequency required to digitally represent a signal is twice the highest frequency present in the input signal, thus the highest frequency component that can be reproduced using a digital system will be half the sampling frequency. Since the average upper bound for human hearing is 20kHz, the minimum Nyquist sampling frequency would be 40kHz, however most systems sample at 44.1kHz, 48kHz or higher to improve audio quality, particularly for high frequency components. Obviously it is desirable to have as high a sampling frequency as possible, but due to bandwidth restrictions a compromise must usually be made for a real-time system.

Most digital wireless systems operate in the unregulated 2.4GHz ISM (Industrial, Scientific & Medical) frequency band. This band actually has a centre frequency of 2.45GHz and bandwidth of 100MHz, allowing use of electromagnetic waves ranging in frequency from 2.4GHz to 2.5GHz. The digital data stream which represents the instrument signal can be transmitted using a number of digital modulation techniques such as Phase Shift Keying or Frequency Shift Keying schemes. Under such schemes, the carrier frequency is periodically phase-shifted, or its frequency changed, respectively, where each discrete phase or frequency employed represents a specific sequence of bits. By analysing the received signal with reference to the original carrier, the bit stream can be reproduced at the receiver. Following decoding if necessary, the resultant sequence of PCM values is passed to a Digital to Analogue Converter, which varies the output level of the receiver to reproduce the quantized instrument signal as obtained by the transmitter.

4.4 Analogue vs Digital Technology

Both analogue and digital wireless technologies have inherent advantages and disadvantages over each other. Since analogue systems require no sampling or quantization of the input signal, they can reproduce the signal with essentially infinite resolution. However, analogue wireless technology is very susceptible to inference from other radio waves, which distort the modulated signal in free space such that the received signal is different from the transmitted signal. This results in unwanted noise at the output of the receiver.

A major advantage of digital technology for wireless communications is that they operate with significantly reduced bandwidth compared to analogue systems. Digital data streams can also be compressed to further reduce the bandwidth requirements of the system. Since digital modulation schemes tend to be more tolerant to free space noise, digital systems are also much less susceptible to radio frequency interference than analogue systems.

However, digital systems introduce quantization error when the sampled signal is approximated to a digitally-representable value, and data between samples is lost. Quantization error places a limit on the maximum Signal to Noise Ratio achievable by a digital system, calculable via equation 4.1.

$$SNR_{\max(dB)} \approx 6.02N + 1.76$$

Equation 4.1 – Maximum Signal to Noise Ratio limit of a digitizing system for sine-wave inputs, assuming uniform distribution of quantization noise, where N is the number of bits used for quantization

The quantization noise can thus be reduced by increasing the bit-depth of the encoded samples (effectively increasing the number of levels for quantisation). The amount of information lost can likewise be reduced by taking more frequent samples. However, increasing the bit resolution or sampling frequencies places greater pressure on the system by increasing the amount of data to be processed. Greater latency is also associated with digital systems, since the processes of sampling, quantization, coding/compression, decoding and demodulation each require a finite amount of time to complete.

4.5 Bluetooth Technology

Widely utilized in mobile phone hands-free systems, laptop and tablet connectivity and wireless audio solutions, Bluetooth is a specification for digital wireless communication maintained by the Bluetooth Special Interest Group. Operating in the same 2.4GHz-2.485GHz band as other standards such as IEEE 802.11 (WiFi), Bluetooth is specifically designed for low-power, short-range applications.

The use of Bluetooth in particular for wireless instrument cable replacement would open up exciting possibilities for wireless connectivity between the instrument and smart devices such as tablet computers, for example the ability to record audio or perform signal processing through a wireless interface. However, traditional problems such as poor audio quality due to bandwidth constraints and latency issues have perhaps rendered Bluetooth technology unsuitable for this application in the past, but recent innovations such as Bluetooth EDR (Enhanced Data Rate) and advanced codecs for stereo audio streaming have served to unlock the potential of Bluetooth technology such that it is now suitable for hi-fi grade audio solutions.

Bluetooth networks are packet-switched, such that communication between modules occurs by the exchange of discrete packages of data. User data is segmented and bundled together with ‘overhead’ data containing information as to, for example, sequencing, priority indication and/or error correction. While this type of communication adds overhead in terms of the volume of data bits to be exchanged, it means that only small portions of data need be retransmitted if a packet gets corrupted. However, for audio transmission, lost or corrupted packets are generally not retransmitted because of the time delay that would be involved.

Since Bluetooth systems use radio frequencies in the same band as many other digital systems including WiFi as mentioned previously, there is a high possibility of interference from other electromagnetic sources, if the system were to operate on a fixed channel (a single carrier frequency). To improve the robustness of Bluetooth networks, and avoid interference to other systems, a technique known as Frequency Hopping Spread Spectrum (FSHH) is employed. Under FHSS, the carrier frequency used to carry information is rapidly switched among a number of channels. The switching sequence is known by both the transmitter and receiver units, allowing the receiver to ‘listen’ only to the specific carrier frequency being output by the transmitter.

The Bluetooth specification comprises a large number of profiles, each of which defines the protocols for operation for a specific application or use-case. All profiles rely on one of two types of link between modules – the Asynchronous Connectionless Link (ACL) or the Synchronous Connection-Oriented (SCO) link. Over an ACL link, packets can be broadcast from transmitter to receiver at any time, and are not guaranteed to arrive at the receiver in the correct order. Packets on an ACL link therefore tend to have greater overhead to ensure that the correct order can be determined at the receiver. For a SCO link, a connection must be established between transmitter and receiver for each transmission, in order to specify the timing parameters required for synchronous communication. SCO data packets are received in the same order as they transmitted, and so have less data overhead. [1]

A number of profiles exist in the Bluetooth specification (v2.1 and later) for audio streaming, which are discussed below:-

- *Advanced Audio Distribution Profile (A2DP)*

  A2DP is used for the streaming of high quality mono or stereo audio, operating over an ACL link. Devices such as stereo headphones commonly couple A2DP with Audio Visual Remote Control Profile (AVRCP) to implement functionality for track skipping and play/pause features.

- *Hands-Free Profile (HFP)*

  HFP is designed for mono-channelled voice audio transmission for built-in vehicular hands-free telephony systems. In contrast with A2DP, HFP audio is transmitted over a SCO link.

- *Headset Profile (HSP)*

  HSP is similar to HFP, but includes the functionality to implement controls such as making, answering or rejecting a call and adjusting call volume. HSP is commonly used for connections between mobile phones and Bluetooth Headsets. Audio is mono-channelled voice data transmitted via SCO link.

It is obvious that the Advanced Audio Distribution Profile (A2DP) is the most, but not the only, suitable Bluetooth profile for the wireless instrument cable replacement application. The A2DP profile specification requires implementation of the Sub-Band Codec (SBC) for audio compression, but does allow for support of non-A2DP codecs, whose operation is to be defined by the manufacturer of a Bluetooth implementation. [2]

5. Audio Quality Assessment

5.1 Audio System Specifications

There are two main groups of phenomena which degrade the quality of audio in any audio system: distortion and noise. Signal distortion is manifested in a variety of different manners, but essentially causes any property of the output signal which differentiates it from the input (other than amplitude gain and noise). In effect, distortion is the alternation of an input waveform such that the output waveform takes a different shape. Distortion is caused by the non-linear characteristic of any audio or communications system and so the actual effects of distortion are dependent on the input signal. While distortion ('clipping' in particular) is often intentionally introduced to a guitar signal to produce a desired sound, any distortion introduced by the instrument signal transmission system is undesired, since its purpose is to carry the signal produced by the instrument's pickups as accurately as possible.

Conversely, noise is generally random in nature and is independent of the input signal. Noise is defined as any unwanted signal that interferes with or impairs the quality of the input signal. Noise can be added to a signal at almost any point in an audio system by any electronic component (thermal or shot noise). In the case of wireless communications, noise can also be added as a result of interference from external radio waves. For audio systems, noise is audible as a low-level 'hiss' or 'hum', particularly when the signal level from the input is low or zero.

There are many quantitative metrics used in assessing the quality of audio equipment, in terms of the amount of distortion and noise that the system introduces to a signal. Some of these are summarised below:-

- *Total Harmonic Distortion (THD)*

  Harmonic distortion is a phenomenon whereby frequencies that are integer multiples of those present in the input signal are added to the signal. For example, for a single input frequency f_1, frequencies of $2f_1$ (second harmonic) and $3f_1$ (third harmonic) and so on will be present at the output. Total Harmonic Distortion (THD) is a measure of the total amount of harmonic distortion (up to a certain harmonic) introduced by the system for a single input frequency. THD is therefore an indication of the tendency of a system to introduce harmonic distortion, where lower THD indicates a higher quality system.

- *Total Harmonic Distortion plus Noise (THD+N)*

Total Harmonic Distortion plus Noise is similar to THD, but considers all components present in the output signal that are not present at the input of the audio system, i.e. both harmonic distortion and noise.

- *Intermodulation Distortion (IMD)*

Intermodulation distortion is caused by interaction between two or more input signals, producing spurious signals which are not harmonically related to the components at the inputs. For two input frequencies f_1 and f_2, intermodulation distortion frequencies such as $f_1 \pm f_2$ (2^{nd} order IMD products) and $2f_1 \pm f_2$ and $2f_2 \pm f_1$ (3^{rd} order IMD products).

- *Signal to Noise Ratio (SNR)*

Signal to Noise ratio is the ratio of desired signal power to the noise power at the output of a system. The higher the ratio, therefore, the better the audio system since the output signal power is large in relation to the noise power, so noise is less audible. SNR is generally measured by finding the noise output for no input signal, comparing that with the output power with an input, and so varies with input signal power. Measuring in this way actually yields a 'signal plus noise and distortion' to noise ratio, and so gives somewhat inaccurate readings since greater output power caused by distortion components actually improve the SNR measurement.

- *Signal In Noise And Distortion (SINAD)*

Signal In Noise And Distortion is a signal quality metric similar to SNR, instead measuring the ratio of the total signal power to the power of all noise and distortion components, i.e. everything that isn't the desired signal. As with SNR, SINAD is expressed in deciBels, where higher values indicate better signal quality. SINAD is arguably a better metric than SNR for audio systems since it takes into account the possibility of distortion components and noise levels varying with input frequency.

- *Frequency Response*

A system's frequency response is the total range of frequencies which can be detected and utilized by the system. This measurement is not an indication of distortion or noise, but relates to the filtering property of the system. For example, a system with frequency response 50Hz-15kHz is considered insensitive to frequencies below 50Hz or above 15kHz; frequencies outside this range are filtered out.

5.2 Qualitative Measurements

There are also a number of techniques recommended by the International Telecommunications Union, the United Nations specialized agency for information and communication technologies, relating to the subjective and objective assessment of audio quality. These techniques are suitable for assessing the impact on perceived quality of lossy audio codecs for digital audio.

5.2.1 Formal Listening Tests for Subjective Assessment

Listening tests are a traditional method of assessing the subjective quality of audio. There are a number of recommendations and standards relevant to this application, but the principal of assessment is essentially the same – to produce a Mean Opinion Score (MOS) by statistical analysis of subjective grades given to audio samples by a large quantity of expert listeners. The MOS scale is a five-grade scale which can be continuous or discrete depending on the method of assessment, and can be used to assess the perceived Quality of audio or Impairments with reference to an ideal sample. The MOS scale is shown in Figure 5.1.

MOS	Quality	Impairment (SDG)
5.0	Excellent	Imperceptible
4.0	Good	Perceptible, but not annoying
3.0	Fair	Slightly Annoying
2.0	Poor	Annoying
1.0	Bad	Very Annoying

Figure 5.1 – Mean Opinion Score Scale [3]

Note that the Impairment scale is also known as the Subjective Difference Grade (SDG), and can be scaled from 0.0 (Imperceptible) to -4.0 (Very Annoying).

The ITU has a number of recommendations for subjective audio quality assessment for a range of applications, including BS.1534-1 which defines the MUSHRA (MUltiple Stimuli with Hidden Reference and Anchor) test. This recommendation is intended for “subjective assessment of intermediate quality level of coding systems”. [4] The method of assessment involves playing a number of audio samples including the original, unprocessed sample and the compressed sample to a panel of expert listeners, who then grade the quality of samples on the continuous Quality scale used to compute a MOS score.

Another ITU recommendation, BS.1116-1, defines “methods for the subjective assessment of small impairments in audio systems including multichannel sound systems” [5] and is also applicable to the assessment of advanced audio coding schemes. Following similar procedures as the MUSHRA test, listeners grade the difference between the original and processed samples using the continuous Impairment scale (SDG) describing the difference as somewhere between “imperceptible” and “very annoying”. Again, the grades obtained from a large quantity of listeners is used to produce a MOS.

5.2.2 Auditory Perception Models for Objective Assessment

The problem with subjective assessments is that they tend to yield different results depending on the listeners’ experience and other factors. The use of auditory perception models, which are computer simulations that attempt to mimic the way in which the human ear hears sound based on psychoacoustic study (discussed later), can be used to obtain an objective measurement of audio quality. Since the model will always produce the same outputs for a certain input audio sample, such measurements can meaningfully be used to compare the quality of different samples.

The ITU recommendation BS.1387-1 defines such a model: the Perceptual Evaluation of Audio Quality (PEAQ) algorithm. This algorithm takes an unprocessed reference sample and corresponding processed audio sample as inputs, computing a number of metrics known as Model Output Variables (MOV) including Disturbance Index and Noise-to-Mask Ratio. These measurements are used to produce a model MOS called the Objective Difference Grade (ODG - as opposed to the SDG).

5.3 Noise Weighting

The fact that the human ear is more sensitive to certain frequencies than others within the audio spectrum can render certain audio system measurements somewhat redundant. If a system produces noise of all frequencies in the spectrum at equal power levels (i.e. white noise), certain frequency components will have a more detrimental effect than other components. Noise Weighting, a procedure whereby each component of the noise is weighted (i.e. multiplied by a nominal value) in an attempt to make noise measurements more indicative of what is heard. Using such weighting a system which produces overall greater noise power shall yield better SNR and possibly THD+N measurements to a system which produces less noise overall but more noise at frequencies to which the human ear is more sensitive.

While there exists a number of noise weighting specifications, the ITU-R 468 standard referring to noise measurements defines the curve which is generally accepted to be the most accurate in terms of modelling the human ear. The standard defines a weighting network to be applied to the device under test, as well as the theoretical response of the network. This response curve defines the weighting values in dB to be applied to the spectrum for noise measurements, where the maximum gain is 12.2dB (relative to 1kHz), applied at 6.3kHz. The ITU-R 468 noise weighting curve is shown in Figure 5.2.

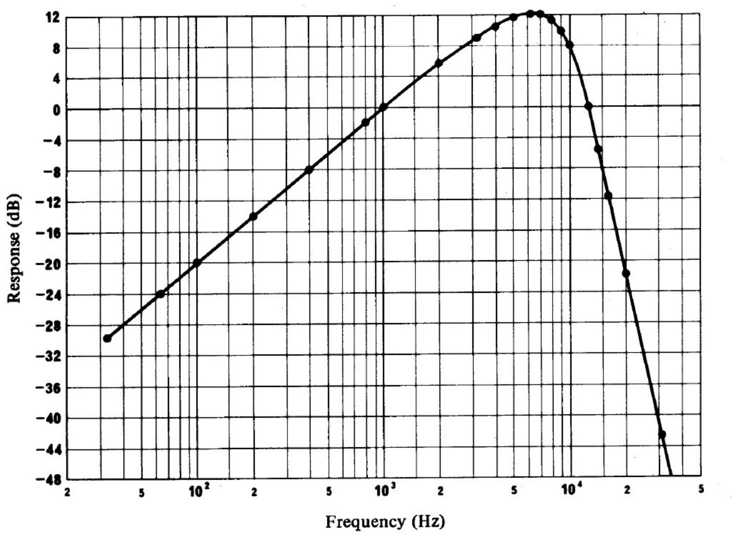

Figure 5.2 – ITU-R 468 Noise Weighting curve [6]

Interestingly, ITU-R 468 noise weighting generally yields worse specifications compared to other methods for noise measurement such as A-weighting or no weighting, so it is rarely used for marketing purposes.

5.4 Impact of Latency

Other than audio quality, latency is another very important factor to consider for audio transmission systems, particularly in the musical instrument application since significantly long delays may inhibit a musician's ability to play. Latency introduced by a wireless instrument cable replacement system is therefore a very significant problem. In fact, one of the main advantages of such a system, the ability to move freely away from the amplifier, can become a liability. This is because as the musician moves away from the audio source, the delay will increase because of the finite speed of sound in air. A generally accepted maximum allowable latency is 40ms, for Audio/Visual applications [7]. However, this limit doesn't necessarily apply to instrument monitoring which also involves tactile feedback as the strings are struck.

A study was conducted by the Audio Engineering Society to determine the effects of latency on live sound monitoring. The study used a MUSHRA testing methodology using a number of musicians to determine a maximum acceptable limit for latency in wedge monitoring and In-Ear Monitoring scenarios for a range of instruments. The study found the acceptable amount of latency to range from 42ms to potentially under 1.4ms under certain conditions. [8] Notably, latency was found to be more acceptable in the wedge monitoring situation even though the actual delay would have been greater than the controlled delay due to the finite speed of sound in air as it travels from the monitor to the ear.

6. Equipment Selection

6.1 Samson Stage 5 Analogue System

The Samson Stage 5 (Figure 6.1) is a versatile analogue wireless microphone system designed to be used with a handheld condenser microphone, hands-free lavalier microphone or guitar or other instrument with a quarter-inch jack output. When using a guitar, the instrument's output is input to the system's transmitter unit via a supplied cable. Claiming greater than 100dB Signal to Noise Ratio, the system operates in the VHF band, using one of 25 frequencies in the range 173.80MHz to 213.20MHz, using Frequency Modulation with 15kHz peak deviation.

Figure 6.1 – Samson Stage 5 wireless transmitter and receiver units (source: www.zzounds.com)

6.2 Joyo JW-01 Digital System

The Joyo JW-01 Digital Wireless Cable Transmitter and Receiver is an inexpensive, rechargeable digital system, operating in the 2.4GHz frequency range, pictured in Figure 6.2. Both the transmitter and receiver unit have quarter-inch jack plugs for direct insertion to the guitar and amplifier respectively. The system is built around the SYNIC IA2E Integrated Wireless Audio Processor and claims total 10ms latency, using a 16-bit codec with 48kHz sampling rate. As such the maximum Signal to Noise Ratio according to equation 4.1 is approximately 98.08dB.

Also included is a microphone input intended to allow transmission of vocals alongside the guitar signal, but the performance of this feature will not be included in testing. The transmitter has an 'anti-interference' button, whose operation is unclear but presumably causes the system to switch wireless channels, indicating that the system does not use automatic frequency hopping, unlike Bluetooth.

Figure 6.2 – Joyo JW-01 transmitter and receiver units (source: www.amazon.co.uk)

6.3 Laird DVK-BTM511 Bluetooth Development Kit

The Laird DVK-BTM511 (Figure 6.3) is a development kit intended for demonstration of Laird Technologies' BTM511 Bluetooth Multimedia module. With support for Bluetooth audio streaming profiles HSP, HFP and A2DP among others; this module is suitable for a number of multimedia applications including, crucially, wireless audio cable replacement.

Based on Cambridge Silicon Radio's BC05 Multimedia chipset, the BTM511 is an implementation of Bluetooth v2.1+EDR and uses a 16-bit codec with 48kHz maximum ADC sampling frequency and 44.1kHz maximum DAC conversion rate (thus 44.1kHz is the maximum rate for both transmitter and receiver operation). The kit is programmed using a set of AT commands which are communicated to the board via serial communications over USB cable.

The module also includes Cambridge Silicon Radio's aptX audio codec for A2DP streaming in addition to the standard SBC codec, making it suitable for testing the impact of advanced codecs on audio quality and latency. Lower latency and 'greater audio transparency' (presumably meaning wider frequency response) is claimed for the aptX codec with respect to SBC [9].

Figure 6.3 – Laird DVK-BTM511 development kit (source: uk.mouser.com)

7. Experimentation Plan

7.1 General Test Scheme

In order to carry out the experimentation required, it will be necessary to utilize several hardware and software based engineering tools. The National Instruments ELVIS board provides an interface for data acquisition (DAQ) and can be used in conjunction with the LabVIEW programming environment to create tools for testing. The function generator feature of the ELVIS board shall be used to produce the input signals required for each test, where a spectral analysis of the systems' outputs can be obtained by performing an FFT of the output signals, allowing computation of the selected metrics. Input signals for all tests will be at an amplitude of 1V peak-to-peak, the typical maximum output of an electric guitar pickup. The root-mean-square amplitude a sine wave is calculated using equation 7.1 below.

$$V_{rms} = \frac{a}{\sqrt{2}}$$

Equation 7.1 – Root-mean-square voltage of a sine wave, where 'a' is the maximum amplitude of the wave (0.5V for a 1V peak-to-peak signal)

Thus the rms amplitude of a 1V peak-to-peak sine wave is approximately 0.35V. Converting to decibels according to equation 7.2, this equates to a power level of -9.03dB.

$$Power_{(dB)} = 20\log_{10}(V_{rms})$$

Equation 7.2 – Power of a sine wave, where V_{rms} is the root-mean-square voltage of the signal

7.2 Test Procedures

Presented here is an experimentation plan detailing the metrics that will be measured and the methods by which the experiments will be done, in order to carry out the Testing/Experimentation task as part of the Project Plan.

- *Frequency Response*

To measure the systems' frequency response, the input signal frequency will be swept from 10Hz upwards to 25kHz using a number of discrete frequency values. The points at which the output power rolls off to 3dB less than the output power for a 1kHz reference input frequency will be defined as the upper and lower bounds of the frequency response. Where the -3dB points do not lie perfectly on any of the applied frequencies during the test, the approximate limit can obtained using linear interpolation.

- *Total Harmonic Distortion*

THD measurement requires a high purity input sine wave of a single frequency. Performing a spectral analysis of the output signal, the power levels of the harmonics can be deduced. Output harmonics will be measured up to the third harmonic for a range of input frequencies in the range up to 6.6kHz, as applied during the aforementioned frequency sweep test. Since the third harmonic for fundamental frequencies greater than 6.6kHz will be more than 20kHz - the upper bound of the audible frequency range - it is not necessary to use any higher frequency input signals. THD will be calculated as the ratio of the root-mean-square of the amplitudes of the harmonics to the amplitude of the fundamental, according to equation 7.3 below.

$$THD_{(\%)} = \frac{\sqrt{V_2^2 + V_3^2}}{V_1} \times 100$$

Equation 7.3 – THD, where V_1, V_2 and V_3 are the amplitudes of the fundamental, second harmonic and third harmonic respectively

- *Signal in Noise And Distortion*

SINAD metrics can also be computed using results from the frequency sweep test. According to equation 7.4, SINAD is defined as the ratio of the total signal power to the power of all noise and distortion components. Calculating the SINAD for constant-amplitude input frequencies across the audible range, the delivered signal quality for each system can be characterised across the range.

$$SINAD_{(dB)} = P_{TOTAL\ (dB)} - P_{NOISE\ (dB)} - P_{HARMONICS\ (dB)}$$

Equation 7.4 – SINAD, where P_{TOTAL}, P_{NOISE} and $P_{HARMONICS}$ are the root-mean-square powers in dB of the total signal, noise components and distortion components respectively

- *Latency*

To determine the latency introduced by each system, a short burst of sine waves at a specific frequency shall be applied to the input of the transmitter modules and compared with the output from the receivers. The simplest input would be a step input, where the latency would be measured as the time between the rising edge of the step at the input and the corresponding rising edge at the output, but since the systems under test are designed for audio signal transmission, the use of a step input may give accurate results. In fact a step input, effectively a momentary infinite-frequency sine wave followed by zero-Hertz sine wave, may not be picked up by the systems since the frequencies involved lie well outside the audible spectrum. It is expected that latency will be quantifiable by measuring the time delay between the end of the first complete cycle of the input signal and the corresponding point at the output.

- *Interference Rejection*

Testing the systems' susceptibility to radio frequency interference will require the generation of interfering RF signals. This can be done using a separate Radio Frequency noise generator, or by creating multipath interference whereby the signals output by the transmitter antenna reach the receiving antenna via multiple, varying-length paths. The latter leads to constructive and destructive interference at the receiver due to the phase difference of the RF signals, caused by the differences in path lengths, and so differences in propagation times, of the received signals. Analysing the output audio signals of the receiver modules in a noisy environment will allow characterisation of the interference-rejection capabilities of each system.

8. Instrumentation and Methodology

8.1 Hardware

The Educational Laboratory Virtual Instrumentation Suite (ELVIS) from National Instruments is a versatile interface for prototyping and experimentation, incorporating a number of digitally controllable measurement instruments. The NI ELVIS II+ model used throughout this project comprises a total of twelve instruments, including an oscilloscope, digital multimeter, function and arbitrary waveform generator and dynamic signal analyser.

The ELVIS board was used primarily as a function generator and Data Acquisition (DAQ) interface to conduct automated measurements on the various wireless audio systems. The function generator (FGEN) is able to provide sine waves within a frequency range of 0.186Hz to 5MHz, both of which limits are well outside the audible frequency range, allowing for measurements across the entire 20Hz to 20kHz band. The maximum output amplitude is rated 10V peak-to-peak, again well above the level required for tests.

In order to both input the generated signals to the transmitter of the system under test and sample the generator output, two separate metre-long 50Ω coaxial cables were used, via a BNC splitter connected to the FGEN output. A BNC-to-quarter-inch mono phone jack attached to the opposite end of one of the coaxial cables allowed for input to the transmitter, using various adapters for compatibility with the different systems to ensure solid, reliable electrical connections. Similar adapters were used on the receiver side, connecting the system's output to an ELVIS analogue input again via a one-metre long coaxial cable (with exposed ends for insertion to the ELVIS breadboard).

Since the source of the test signal is generated by the ELVIS Function Generator, whose negative terminal is internally connected to ground, the analogue input channel for the source signal was set up to make referenced single-ended measurements. As such, each sample is read as the voltage of the positive terminal (connected to the FGEN output and the transmitter input) with reference to the ELVIS's ground.

Configuration of the analogue input channel for the output of the receiver required further consideration. Electric guitar amplifier inputs have a typical input impedance of 1MΩ, while the ELVIS board's analogue input channels are rated at greater than 10GΩ. In order to properly simulate an amplifier input to enable maximum power transfer, then, a 1MΩ resistor was connected across the analogue input terminals for the device under test's output channel. The channel was set up for differential measurement, i.e. the voltage reading for each sample is taken as the difference between the positive and negative terminals of the receiver's output. A wiring schematic for the full measurement setup is presented in Figure 8.1.

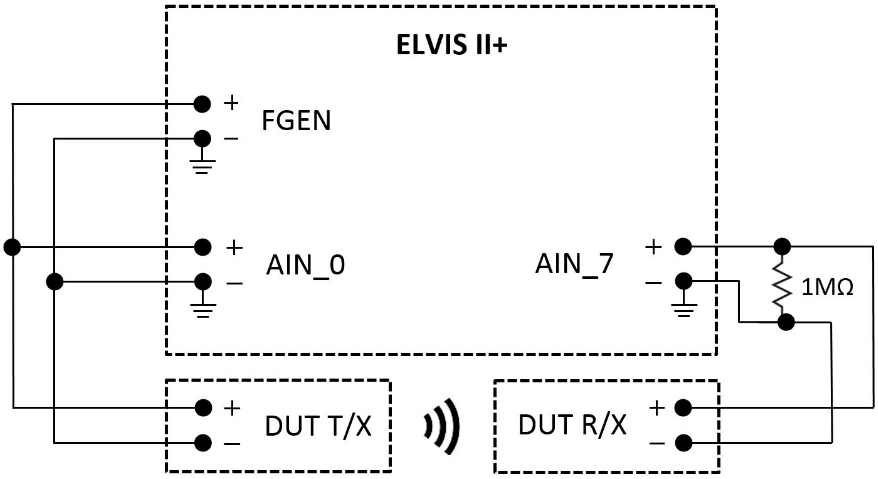

Figure 8.1 – ELVIS measurement setup Schematic (FGEN - Function Generator, AIN - Analogue Input, DUT – Device Under Test)

8.2 Software – Data Capture and Test Automation

The functionality of the ELVIS board can be extended through the development of proprietary Virtual Instruments using the LabVIEW programming language and environment. The creation of such VIs also allows for automation of test procedures for ease and consistency of experimentation. The programs produced for use in this project were developed using a Data Flow Design methodology, where functions of the software are described in terms of the flow of data and the transformations that are performed on the data. The sampling rate used is 50,000 samples per second, yielding a Nyquist measurement bandwidth of 25kHz, while allowing adequate settling time between samples for maximum accuracy. Anti-aliasing filtering is deemed unnecessary due to the fact that all systems are designed to output only frequencies within the audible frequency spectrum, and so any components above 25kHz will be at such a low level that their aliasing effect will be minimal.

LabVIEW programs are made up of two main components; the block diagram, which determines the function and 'inner workings' of the program, and the Front Panel which serves as the User Interface. Front Panels consist of controls and indicators; the former allowing for User input to the program and the latter to display output data in a variety of formats. The Front Panel designs for each program will be presented here along with descriptions of functionality, while Block Diagram program listings for all VIs are included as appendices.

8.2.1 Frequency Sweep VI

Many performance specifications can be deduced by analysing the output waveform of a system for a known input waveform. Performing a frequency sweep across the full audible range allows for comprehensive characterisation of the system. The Front Panel of the Frequency Sweep VI, shown in Figure 8.2, includes controls for the amplitude of the source signal, sampling frequency and number of samples and FFT window parameter for the frequency-domain displays. Since the frequency of the generated signals are determined automatically at runtime, it is not user-controllable, although the Front Panel does display the current frequency via an indicator. A further control allows the User to specify the file path for the sampled waveform data, opening a dialogue box if left empty at completion of the program.

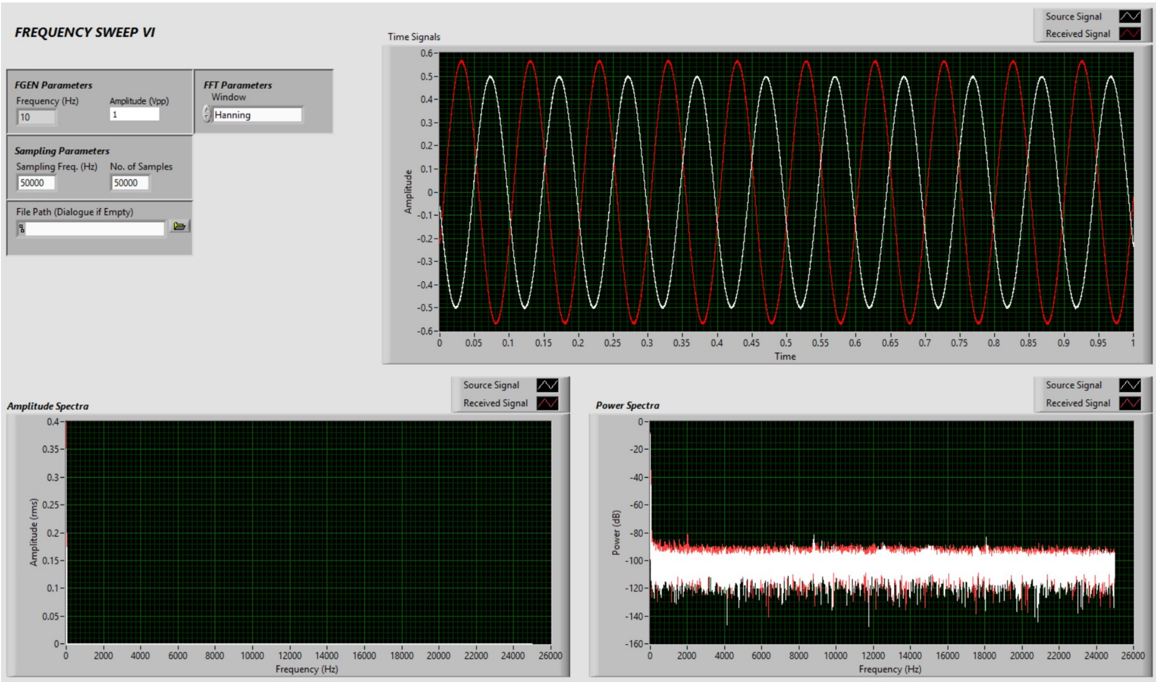

Figure 8.2 – Frequency Sweep VI Front Panel showing the display after applying a 10Hz input signal to the Joyo JW-01 system

The program is set up to apply sine waves from 10Hz – 25kHz to the input of the wireless system at intervals of 10Hz up to 100Hz and of 100Hz thereafter. The increased resolution at the lower end of the spectrum reflects the phenomenon that the systems' response since the response tends to roll of much more quickly at the lower end than at higher frequencies, allows for accurate characterisation of each system's frequency response. Prior to taking any samples, the Function Generator is run for one second, to prevent any spurious signals produced on startup from obscuring the resultant output signals from the wireless system. Samples are then taken of both the input and output channels for a further one second. Once the samples are acquired, the Function Generator is stopped and the acquired data is presented on-screen in three formats; a simple time domain waveform, and amplitude and power spectra in the frequency domain. [Note that these displays are for monitoring purpose only and are to be regenerated by a separate VI at a later stage for analysis.] On completion of the entire frequency sweep, the sampled waveforms are saved to storage as a file, to be opened by a separate VIs for analysis. The sequence of execution carried out by the Frequency Sweep VI is summarised in the Flow Chart of Figure 8.3.

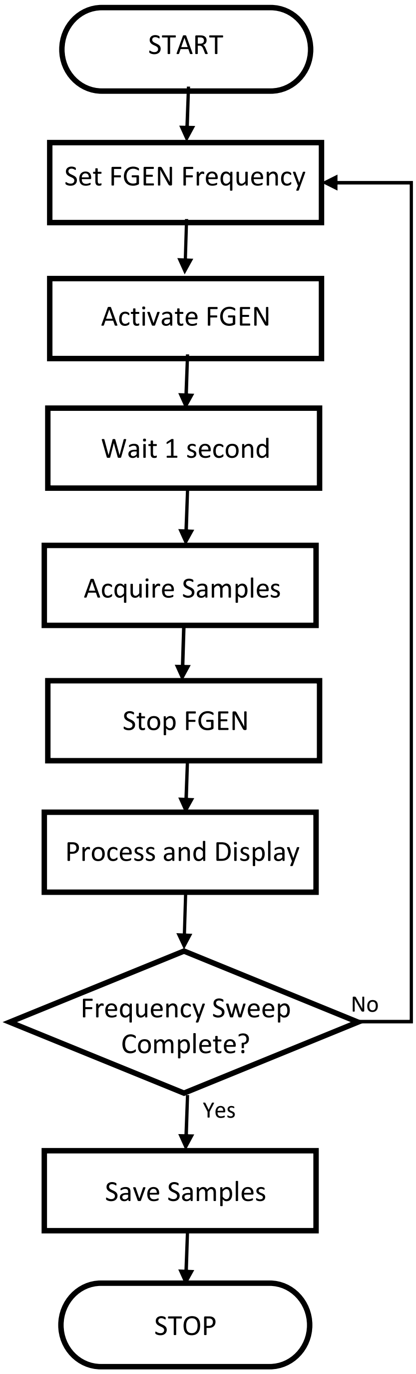

Figure 8.3 – Flowchart illustrating the operation of the Frequency Sweep VI

8.2.2 Latency VI

The Latency VI is designed to apply a burst of sine waves to the input of the device and sample both the input and output signals. The data captured is again to be analysed later and the end-to-end latency of each system determined. The front panel of this VI, as shown in Figure 8.4, includes controls for setting the frequency, amplitude and burst duration of the source signal. Also included are a Sampling Frequency control and 'Wait' control, the latter of which is used to determine the number of samples to be taken before and after the burst, to ensure that the entire input and output signals are sampled.

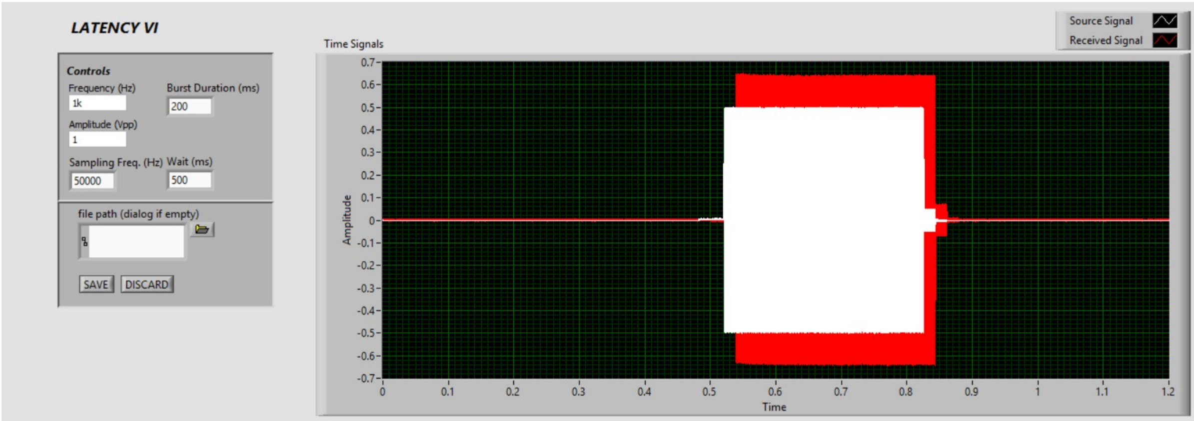

Figure 8.4 – Latency VI Front Panel, showing the display after applying a burst of 1kHz sine waves to the Joyo JW-01 system

8.3 Software – Analysis Tools

8.3.1 Audio Measurements VI

The Audio Measurements VI is designed to take as input the waveform files as produced by the Frequency Sweep and Latency VIs. The program parses the information in the input file to reconstruct the sampled waveforms, re-displaying the time signals on the Front Panel as they are read in. For each pair of waveforms contained in the input file, i.e. for each frequency employed under the Frequency Sweep test, the Fourier Transform is applied to produce both amplitude and power spectra in the frequency domain, applying a Hanning window (discussed under Chapter 10 *Discussion*). For input frequencies up to 6.6kHz, the THD ratio is computed and displayed on a graph, as is the SINAD measurement across the full 10Hz – 25kHz range. Plotting the power of the fundamental against frequency for each output signal, the systems' frequency response is also drawn. The Front Panel of the Audio Measurements VI is shown in Figure 8.5.

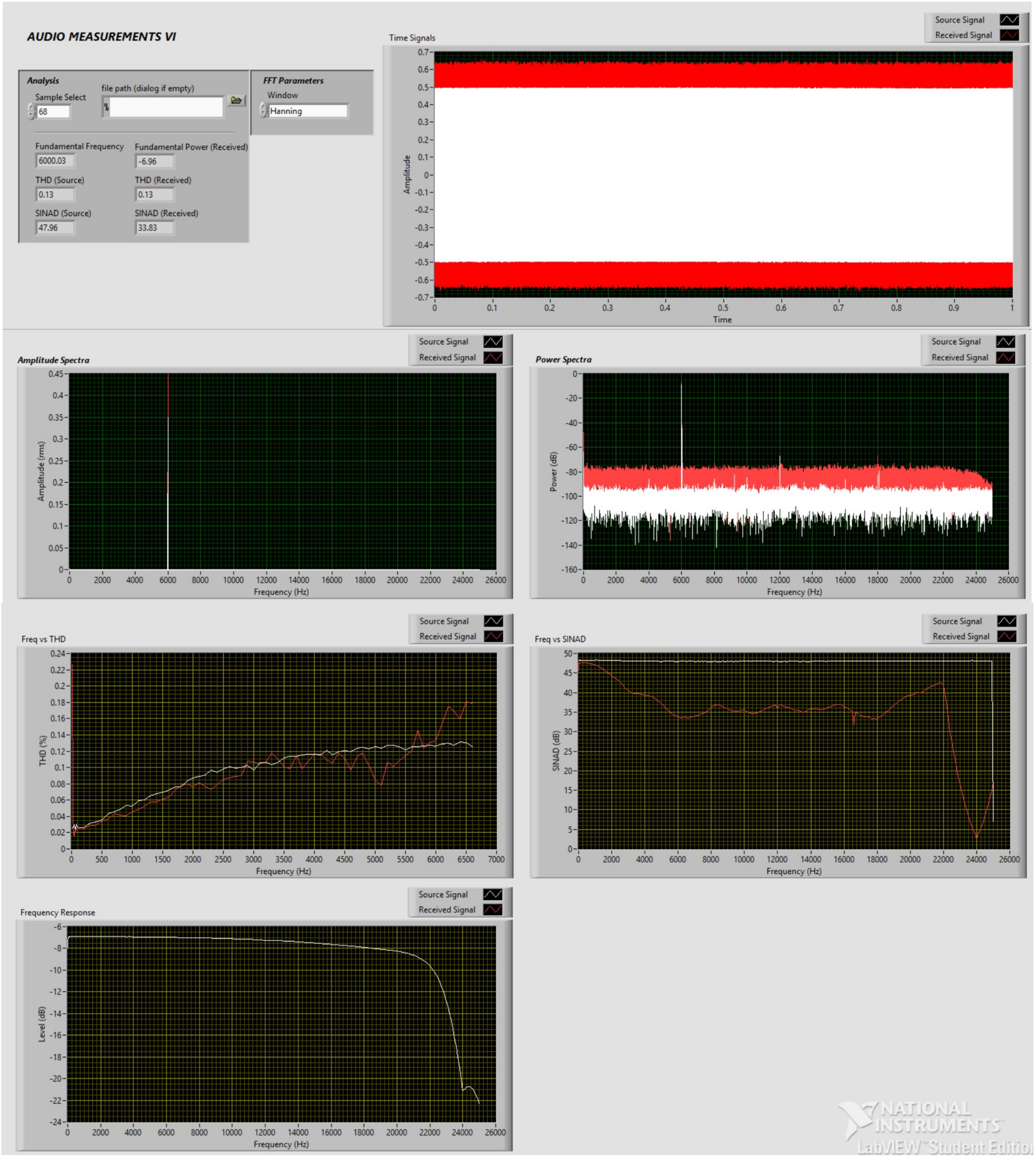

Figure 8.5 – Audio Measurements VI Front Panel, showing the display for the Joyo JW-01 Frequency Sweep data, with the 6kHz input samples selected

Following computation of the metrics as described above, the user is able to navigate through all the sampled waveforms contained in the input file, where the time signals, power spectra for those particular samples are displayed. Also shown is the detected fundamental frequency and respective power (in dB), and THD and SINAD measurements, where applicable, for both the stimulus and response signals. This intuitive interface allows for quick, yet detailed analysis of all the sampled signals obtained during data acquisition.

Data produced by the Audio Measurements VI can be exported to Microsoft Excel for further computation, such as graph generation and interpolation as necessary for determination of latency and Frequency Response limits. All graphs displayed in Chapter 9 *Test Results & Analysis* are produced in Excel using data imported from the Audio Measurements VI.

8.3.2 Noise Weighting VI

While entirely possible, it is impractical to construct a physical ITU-R 468 weighting network for noise-weighted measurements, and would require duplication of all measurements with the network applied. It is much more practical and intuitive to post-process the sampled waveforms to apply the required weighting via software. While LabVIEW does support ITU-R 468 weighting, this requires NI's Sound & Vibration Toolkit which was unfortunately unavailable for the project. For the same purpose, a simple MATLAB script was written to design and apply an ITU-R 468 weighting filter to sampled waveforms, which was integrated into a LabVIEW VI. (This script can be seen in the block diagram for the Noise Weighting VI, included as Appendix 4.)

The Noise Weighting VI takes waveform files as output from the Frequency Sweep VI, reformatting them into MATLAB-compatible signal and passing them to MATLAB for processing. The integrated MATLAB script then characterises and applies the noise weighting filter, passing the weighted signal back to LabVIEW. Once all signals present in the input file are processed, the VI repackages them in the LabVIEW format and saves them as a separate file. Since there is no requirement for the User to visualise the processed data at this stage or make any input other than the input and output file paths when prompted, the Front Panel for this VI is essentially redundant. The filtered waveforms as output by the Weighting Filter VI can be processed in the same way as those unfiltered waveforms by the Measurements VI.

In order to prove the accuracy of the MATLAB-based weighting filter, an additional VI was created to apply a white noise signal to the MATLAB script. Subtracting the frequency-domain power levels of the original signal from those of the filtered signal, the response of the filter was obtained. This response is shown in Figure 8.6.

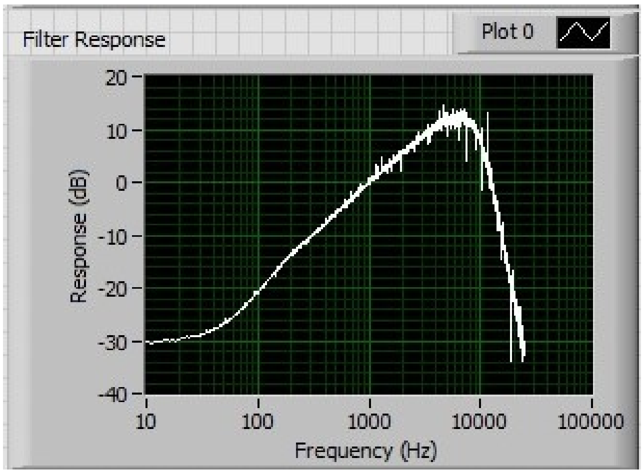

Figure 8.6 – MATLAB designed ITU-R Weighting Filter Response

The MATLAB filter response (Figure 8.6) is shown on a LabVIEW Front Panel indicator, plotted on a logarithmic scale for easy comparison with the actual ITU-R 468 curve of Figure 5.2. It can be seen that the filter appropriately models the ITU-specified curve.

8.4 System Setup

In order to ensure consistent conditions for each system under test, like output interfaces were employed as available and the overall gain of each system tuned as close to unity as possible, for a 1kHz input signal. Setup considerations for each system are discussed here.

The Samson Stage 5 features both a balanced XLR-type output and an unbalanced line-out interface for audio output. In order to ensure consistency with the other systems, which do not have a balanced output, the line-out jack socket was used for all measurements. The overall gain of the system is adjustable via knobs on both the transmitter and receiver modules. In order to ensure consistency throughout all tests, only the gain control on the receiver module was adjusted to give approximately unity-gain for a 1kHz input, and the position of the knob noted and kept uniform for all tests.

The gain of the Joyo JW-01 system is not adjustable, and so unity-gain could not be ensured for this system. The only setup required for this system was to pair the transmitter and receiver modules on initial power-up, according to the instructions provided with the system.

The Laird DVK-BTM511 Bluetooth development board required setup via the communication of commands via serial interface over USB from a PC. Setup, pairing and codec activation was performed by issuing serial commands according to instructions provided in the User Manual for the development kit. The gain of the system was set to approximately unity for a 1kHz input, by sending commands to the receiver module in order to tune its output gain.

8.5 Environments

The systems have been characterised under a number of different environmental conditions, which typically effect wireless communications performance in a number of ways. While an ideal wireless environment would be a large open space, with no obstacles or interfering equipment or signals present, it was not practical to access such an environment. Such conditions are also not typical of the environment where a wireless instrument cable replacement system would be used, and so were deemed unnecessary under the scope of the project.

8.5.1 Typical Conditions

The first conditions under which the systems were tested was a typical, fairly ideal wireless environment. The receiver and transmitter were positioned at a distance of one metre apart, measured using a metre rule, on a desk either side of the ELVIS board in a medium-sized laboratory. The measurement setup is shown for the Joyo JW-01 system in Figure 8.7.

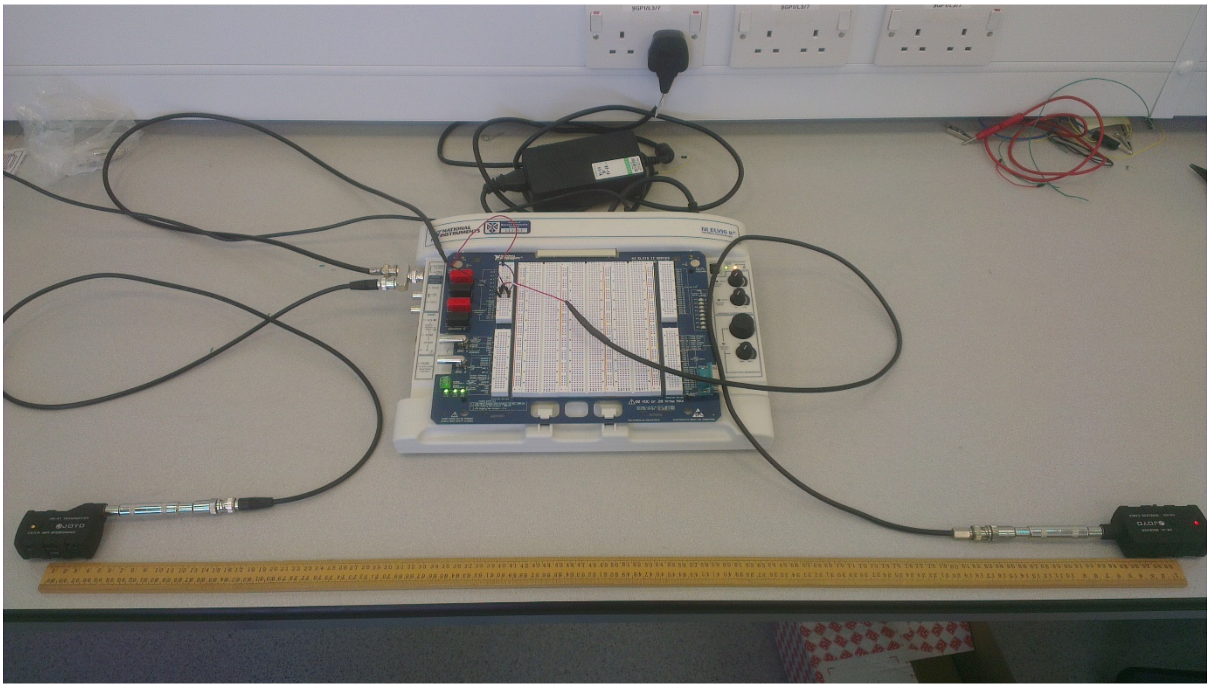

Figure 8.7 – Measurement setup under typical conditions, showing the Joyo JW-01 system under test

While it was not possible to completely ensure sole occupation of the 2.4GHz ISM band for the digital systems, precaution was taken to clear the band as much as practically possible by disabling the communication functionality of all nearby Bluetooth- and WiFi-enabled devices. This minimised the probability of interference between the systems under test and those in the environment as a result of ongoing data transfers or heartbeat messages (e.g. between a WiFi router and connected device). The same precautions were not ensured during the analogue system tests, since the frequencies used for digital communication are well outside this system's frequency band, and so will not interfere.

8.5.2 Non-Ideal Conditions

Non-Ideal condition tests were conducted inside a Reverberation Chamber, purpose-built for testing wireless communication system performance. The chamber consists of a Faraday cage screened room with programmable moving reflective paddles for tuning the conditions within the chamber. For all tests conducted under this project, the paddles were set to continually sweep along the full width/length of the chamber, creating continually fluctuating RF conditions to cause randomised disturbance to the systems under test.

The main effect of the reverberation chamber is to reflect RF signals continuously, such that they reverberate around the environment for an extended period. This creates multipath links between the transmitter and receiver, i.e. the transmitted signal can follow multiple paths to reach the receiver. Since each path differs in length, the signal at the receiving antenna will consist of a number of signals out-of-phase with each other, causing constructive or destructive interference. In addition, since signals bounce around the chamber for a while after transmission, 'old' signals will also affect the received signal. As such it is to be expected that the audio signal quality at the output of a wireless system under reverb chamber conditions will be degraded, or digital links will be lost as packet data is distorted.

The distance between transmitter and receiver, while not precisely measured, was regulated so that it was kept uniform for all tests by marking the approximate antenna positions using tape and ensuring that for all tests the modules were in line with these markers. A total of three separate experimental environments were set up within the reverberation chamber;

- Line-Of-Sight Tests, such that a direct line-of-sight path, in addition to the multipath lines induced by the chamber, was existent;
- Non-Line-Of-Sight tests, where the direct line-of-sight path was blocked by the use of a conductive sheet of metal placed in front of the transmitter;
- Phantom Attenuator tests, where the transmitters were held in direct contact with a human tissue-simulating 'Phantom' device. The Phantom absorbs electromagnetic energy, converting it to heat, similar to human tissue.

Under these conditions, a frequency sweep was conducted on each system. The effect of the adverse environment on the performance of the systems is manifested by the shape of the Frequency Response curve as produced by the Audio Measurements VI. The entire frequency sweep procedure takes about 20 minutes, and any dropouts or loss of signal purity which occur during the test show up on the Frequency Response curve as sharp fluctuations from the expected trend. As such, the frequency axes can be read as time, indicating the ability of the systems to cope with multipath interference across an extended time window, rather than according to the input frequency.

Figures 8.8, 8.9 and 8.10 show the setup inside the reverberation chamber for the Laird DVK-BTM511 system under the Line-Of-Sight, Non-Line-Of-Sight and Phantom tests respectively. As previously mentioned, the setup was kept as similar as possible for each of the other systems in terms of positions of the transmitter and receiver modules.

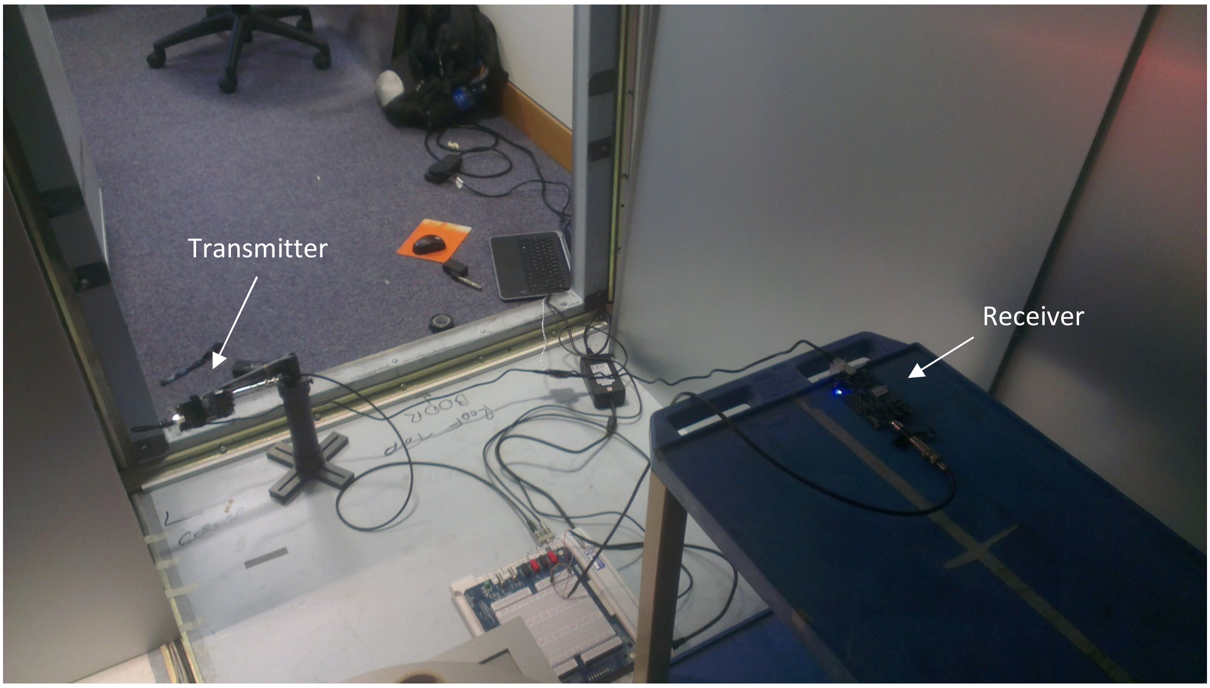

Figure 8.8 – Line-Of-Sight test setup for the Laird DVK-BTM511 Bluetooth system, taken from inside the reverberation chamber

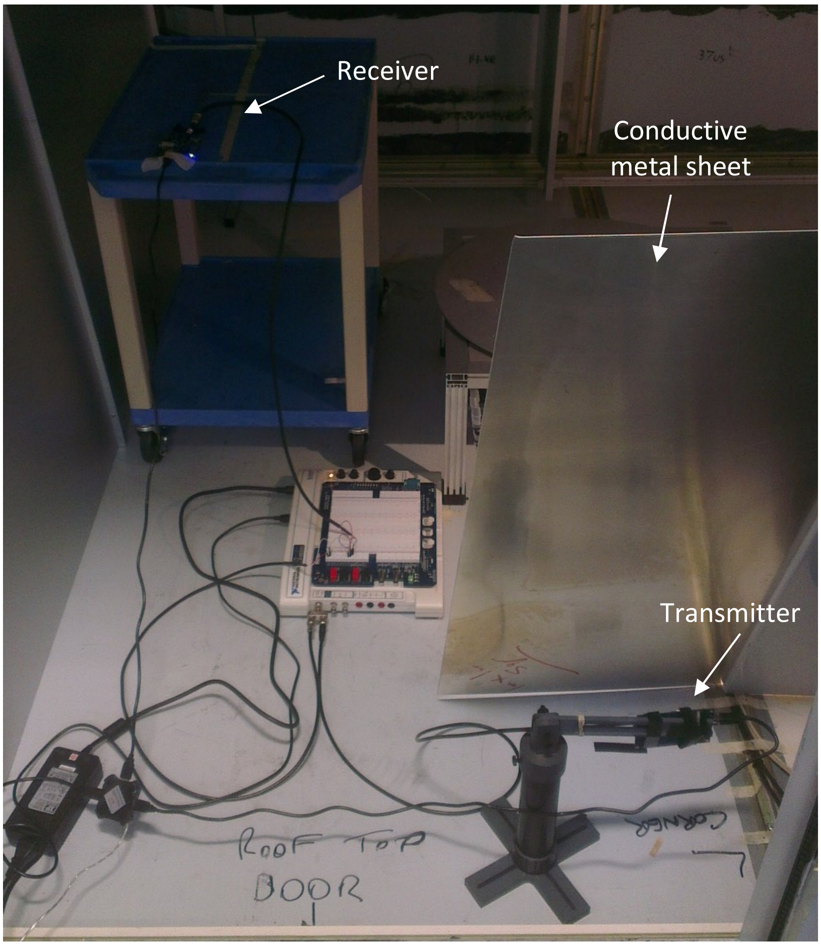

Figure 8.9 – Non-Line-Of-Sight test setup for the Laird DVK-BTM511 system, taken from outside the reverberation chamber door

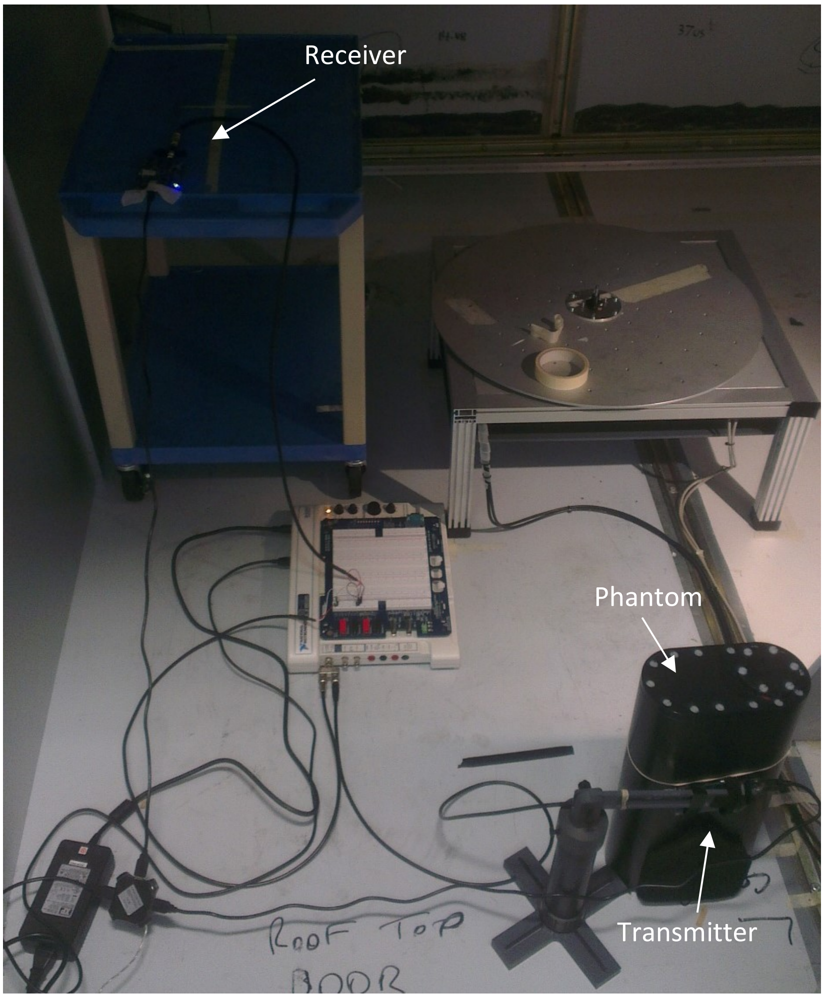

Figure 8.10 – Phantom test setup for the Laird DVK-BTM511 system, taken from outside the reverberation chamber door

It is important to note that due to the requirement to provide power to the ELVIS board and Samson Stage 5 receiver, and USB connections to the Bluetooth development boards, it was not possible to fully close the door of the Reverberation chamber; it had to be kept ajar as an entry point for the necessary cables. While this would have an effect on the performance of the chamber (e.g. allowing some RF signals to escape the chamber), it was a constant part of the environment and so its effect will have been uniform for all tests.

9. Test Results and Analysis

9.1 Samson Stage 5

9.1.1 Frequency Response

The frequency response curve for the Samson Stage 5 system is shown in Figure 9.1 below.

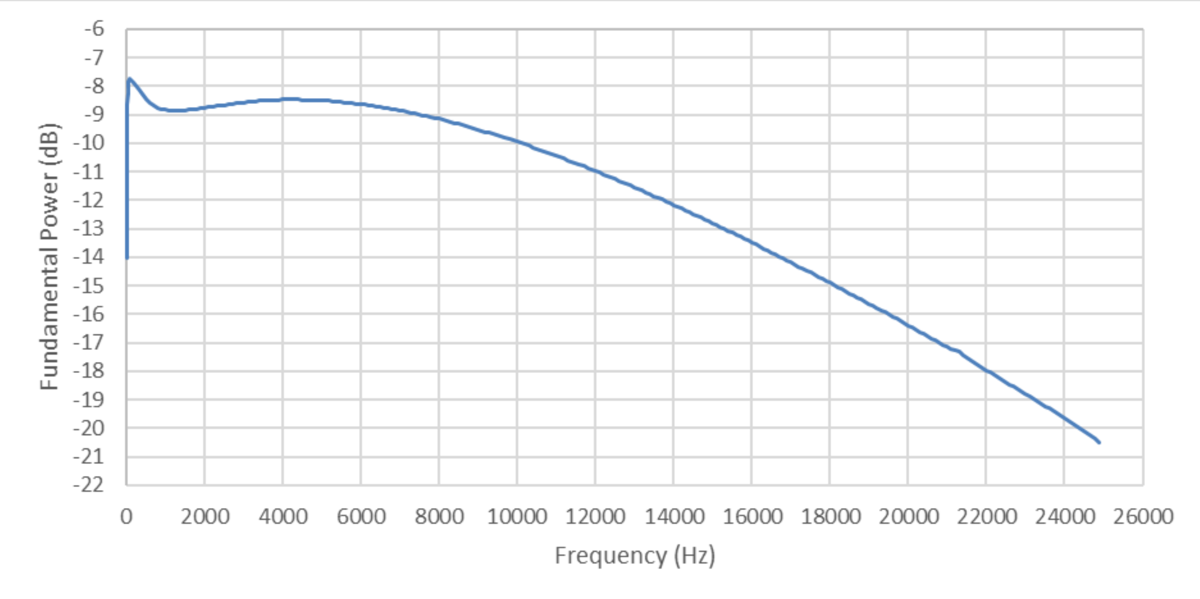

Figure 9.1 – Samson Stage 5 Frequency Response curve

As shown in the above curve, the Samson Stage 5 analogue system has a very non-flat frequency response. Peaking at -7.75dB (representing 1.28dB gain, or 1.15 as a ratio) at just 100Hz, the response dips to -8.86dB at 1300Hz before rising again to -8.47dB at 4400Hz. Beyond this, the response exhibits a steady roll-off of about 6dB/octave (from -10dB at 10kHz to about -16dB at 20kHz). This indicates a simple first-order low pass filter response. At the reference frequency of 1kHz, the response is -8.83dB (0.2dB or 1.02 gain). Relative to this level, the lower -3dB point is 14.97Hz and the upper is at 13485Hz.

9.1.2 Harmonic Distortion

The Harmonic Distortion characteristics of the Samson Stage 5 response and stimulus signal are shown in Figure 9.2.

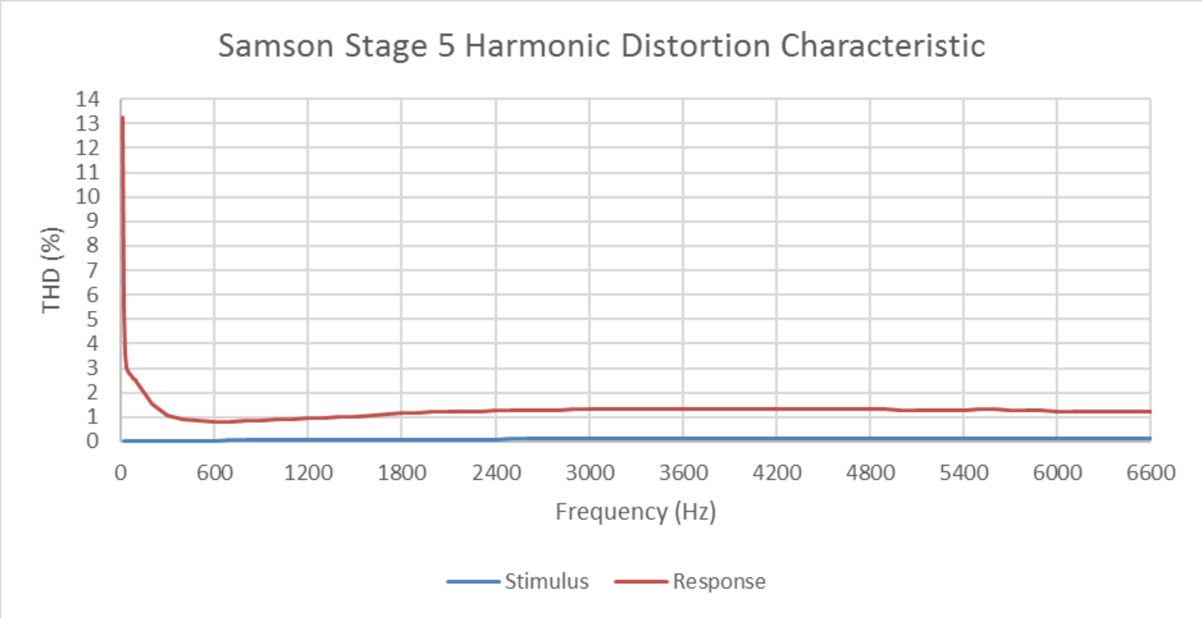

Figure 9.2 – Total Harmonic Distortion vs Frequency for the Samson Stage 5 output and corresponding input

Low input frequencies under about 250Hz are most susceptible to harmonic distortion under this system, with a THD content of greater than 1.5%, while those in the range 300Hz to 1.3kHz are least susceptible; harmonic distortion components accounting for less than 1% of the output signal. Beyond 1.3kHz to the maximum measurable frequency of 6.6kHz, harmonic components remain at less than 1.2%.

Figure 9.2 shows harmonic distortion to be very high for low fundamental frequencies; above 3% of the total output energy below 40Hz. This poor performance is amplified by the fact that, at such low frequencies, the fundamental output power itself is low, as demonstrated by the frequency response curve for this system (Figure 9.1). Figure 9.3 below shows a section of the power spectra for the stimulus and response signals at an input frequency of 40Hz; the peaks at 80Hz, 120Hz, 160Hz and so on being due to harmonic distortion. Note that the input signal shows comparatively very little harmonic distortion.

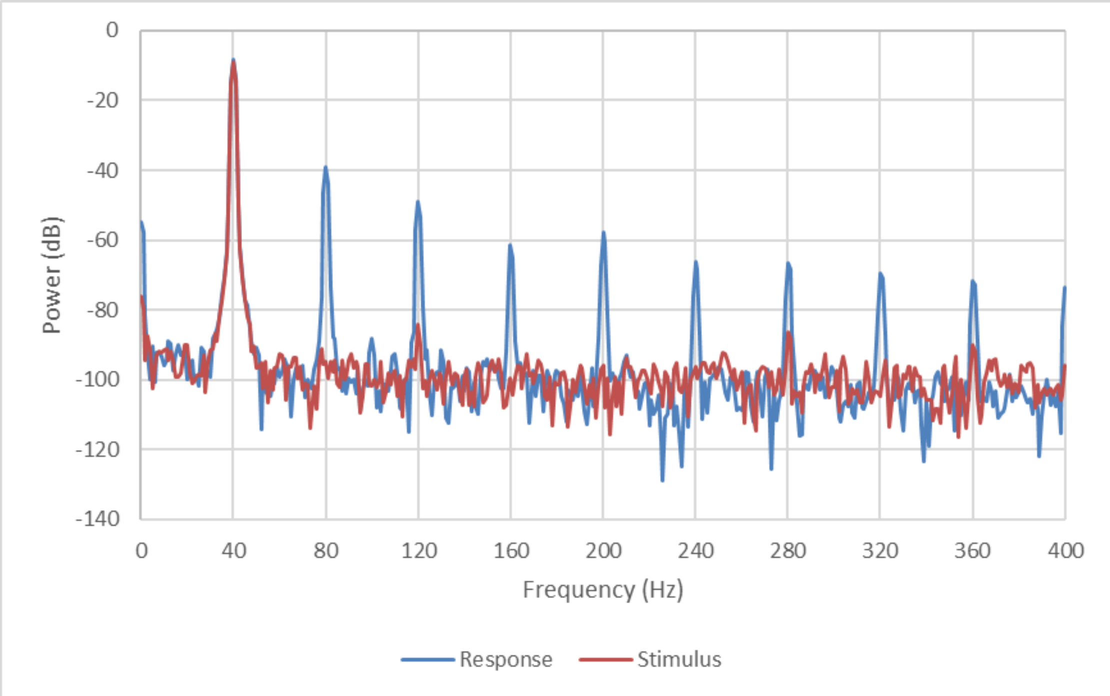

Figure 9.3 – Section of the power spectra of the stimulus and response signals for the Samson Stage 5 at a test frequency of 40Hz, demonstrating harmonic distortion components at the output

As Figure 9.3 above demonstrates, many harmonic tones are produced by the system, beyond the tenth harmonic at 400Hz for a 40Hz input. Thus the actual THD measurements would be higher than those calculated under this test, which considers only the second and third harmonics.

9.1.3 Signal Quality

The Samson Stage 5's measured Signal In Noise And Distortion characteristic for input tones across the audible frequency range is shown in Figure 9.4.

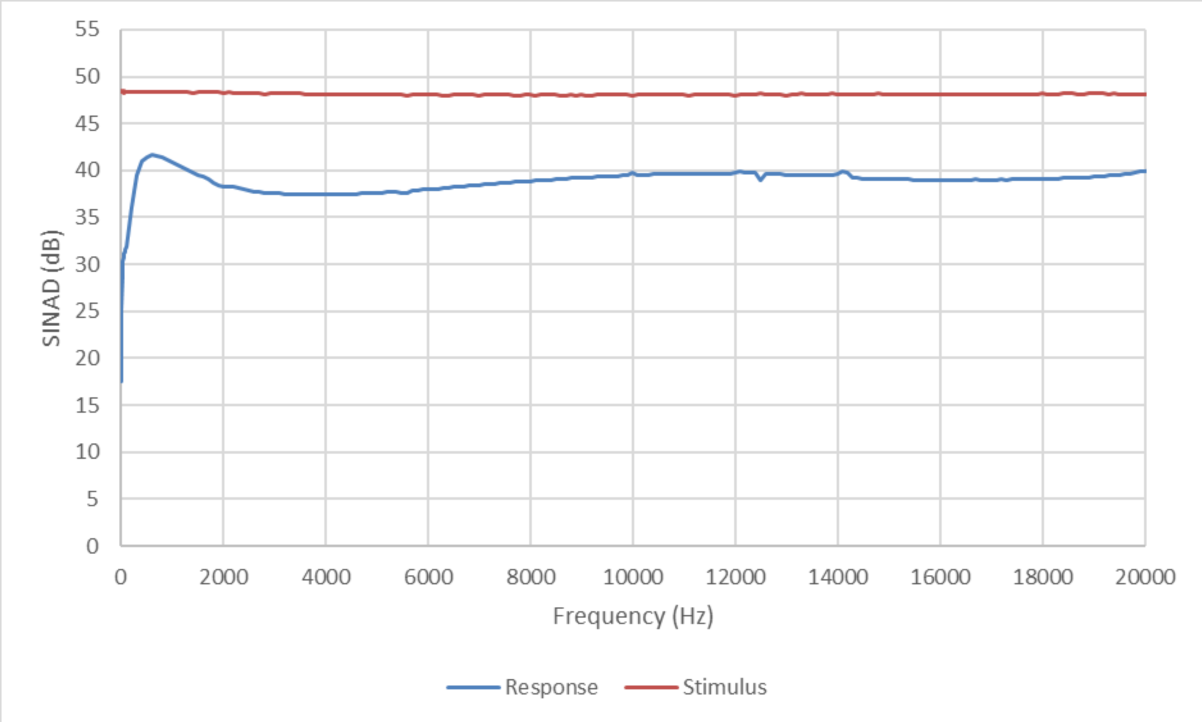

Figure 9.4 – SINAD vs Frequency for the Samson Stage 5, showing both stimulus and response measurements

Analysing the SINAD characteristic for the Samson Stage 5 response and associated stimulus signal, it can be seen that the signal quality remains fairly constant across much of the audible range. The initial section of the response curve on Figure 9.4, from 10Hz – 6600Hz, can be seen as an inversion of the response Harmonic Distortion curve of Figure 9.2; the peak at 600Hz correlating with the minimum THD point at the same frequency. Elsewhere the degraded SINAD measurement with respect to the stimulus signal is due mainly to noise, and the fact that the fundamental output power falls as the input frequency rises, as demonstrated by Figure 9.1.

Maximum SINAD occurs when THD is at a minimum at 600Hz, with a value of 41.62dB. The lowest measured SINAD ratio is 37.41dB at 3900Hz. The stimulus signal remains at about 48dB SINAD across the range.

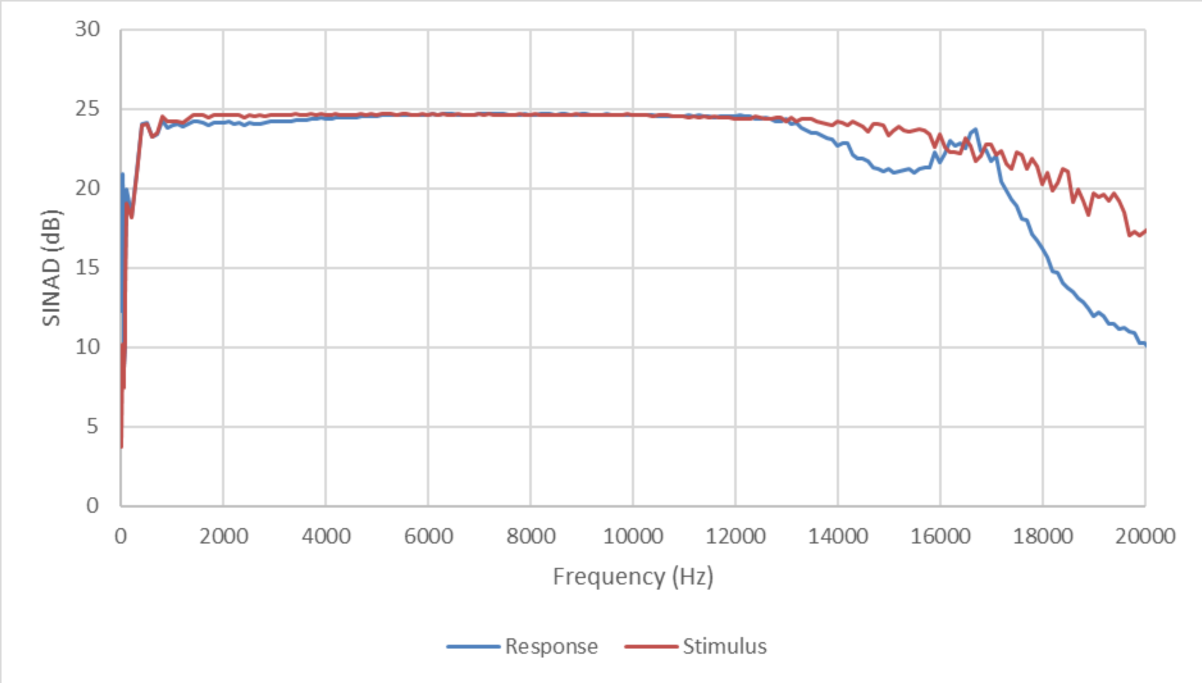

Figure 9.5 – SINAD vs Frequency for the Samson Stage 5, showing both stimulus and response measurements, under ITU-R 468 Noise Weighting

On application of the MATLAB-designed ITU-R 468 Noise Weighting filter, SINAD measurements for both the stimulus and response signals are dramatically worsened, as shown on the SINAD characteristic graphs of Figure 9.5. This is due to the effect of the filter which boosts frequencies around 6kHz as discussed under Chapter 5.3 *Noise Weighting*. The filter has also had the effect of changing the shape of the Stimulus SINAD characteristic graph, resulting in lower quality signals at the extreme ends of the frequency range. This, again, is due to the response of the filter, lowering the fundamental power with respect to the noise power elsewhere in the spectrum.

9.1.4 Latency

Applying an input signal burst at 1kHz fundamental frequency, the response at the output of the Samson Stage 5 receiver is as shown alongside the corresponding input on Figure 9.6.

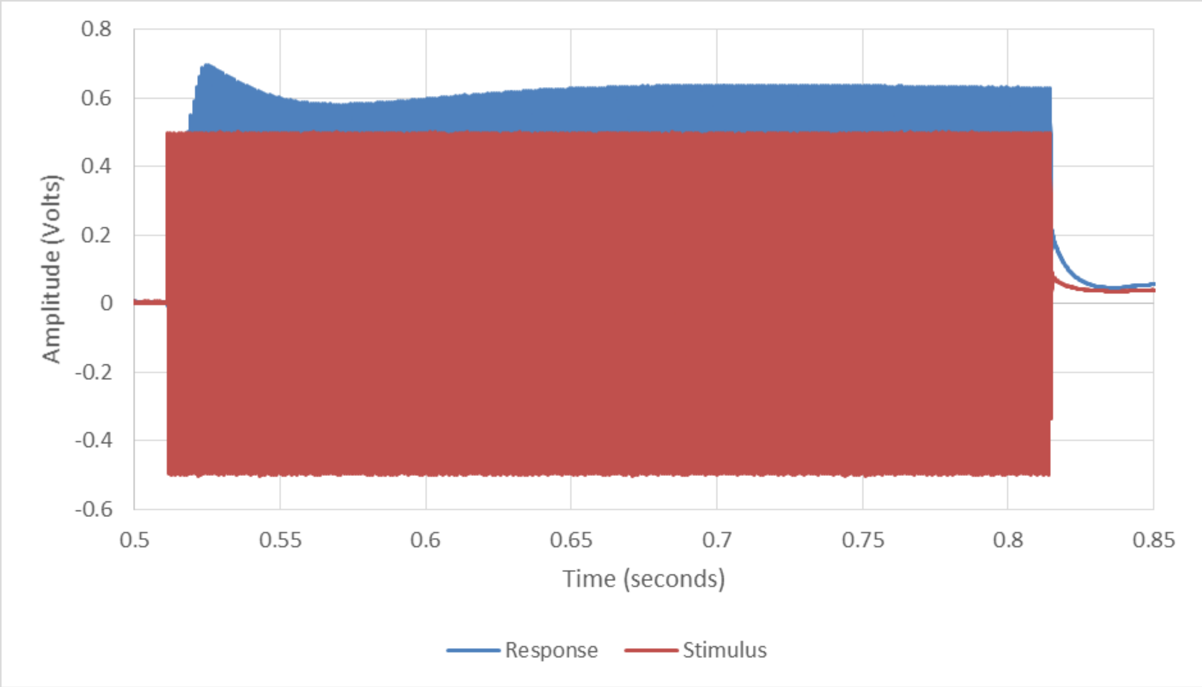

Figure 9.6 – Stimulus and response waveforms for the Samson Stage 5 under the Latency test using a 1kHz frequency

The initial few cycles of the waveforms are more clearly seen on Figure 9.7.

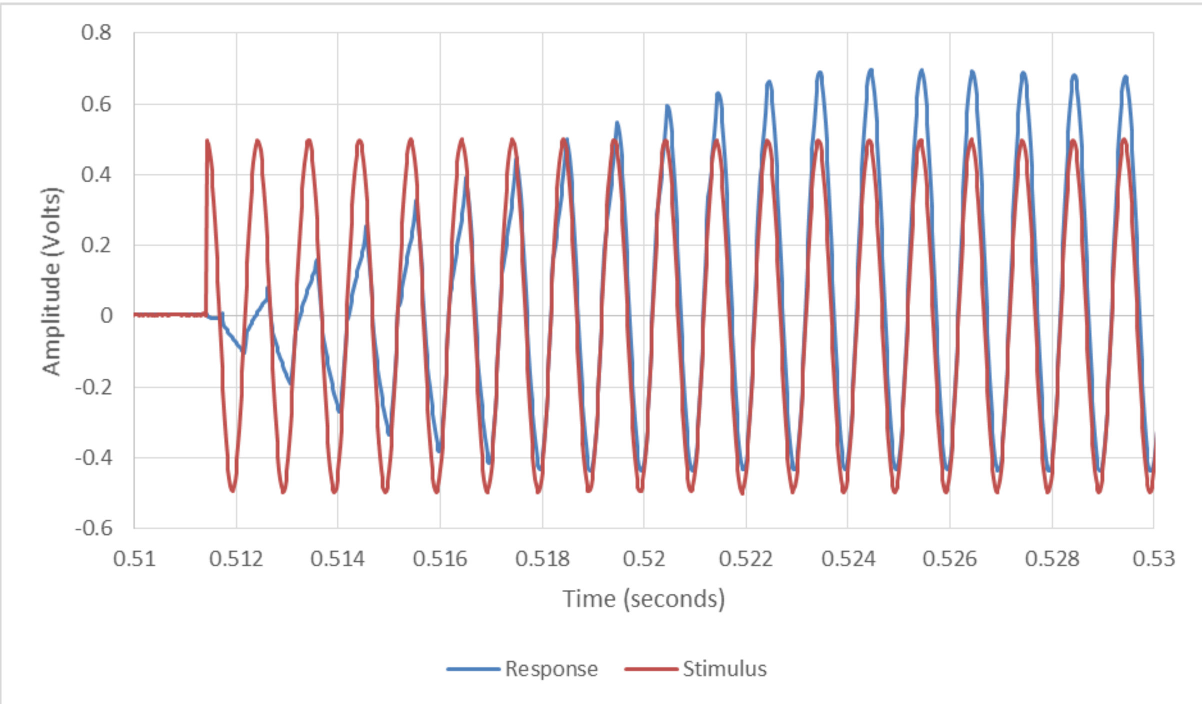

Figure 9.7 – Stimulus and Response waveforms for the Samson Stage 5 under the latency test using a 1kHz frequency, showing the initial few cycles of the bursts

Looking at the initial few phases of the source and received signals from the Samson Stage 5 under the Latency Test in Figure 9.7, it can be seen that the analogue system induces minimal latency in terms of time taken to see an output signal following the input. In fact, the output waveform remains effectively in phase with the input signal. However, the system shows a rise-time characteristic typical of analogue systems. This can be more clearly seen on Figure 9.8, which shows only the initial section of the output waveform of the system under the Latency Test.

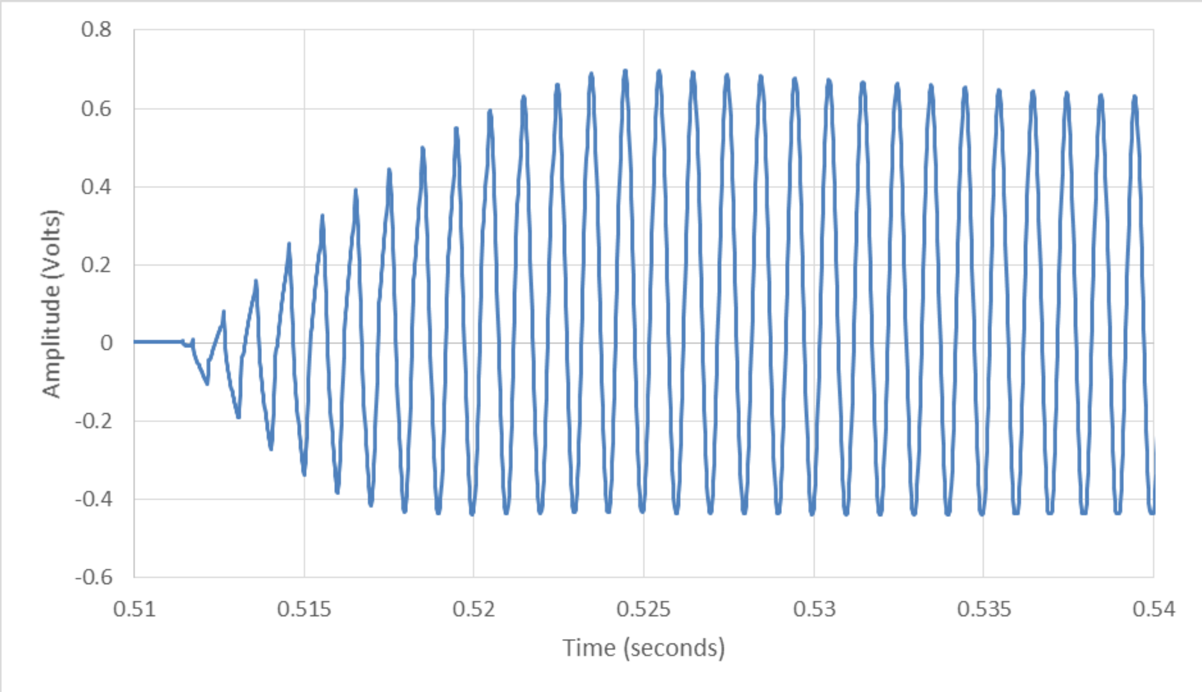

Figure 9.8 – Initial section of the response waveform for the Samson Stage 5 under the latency test using a 1kHz frequency

As mentioned above, the system exhibits a rise time characteristic where the amplitude of the initial phases of the output waveform rise gradually from essentially zero to a maximum level before trailing off to a steady amplitude. This phenomenon is analogous to the response of an analogue control system to a step input, which will similarly rise from zero to a peak overshoot value, then tend towards steady-state.

A problem arises when trying to quantify the latency induced by the system, since the response is so different to what would be expected from a digital system. While an output is produced virtually immediately on application of an input, it can be seen from Figure 9.8 that the initial cycles of the response are low-amplitude, low-quality sine waves which cannot be taken as appropriate reproduction of the source signal.

Figures 9.9, 9.10 and 9.11 show the captured response waveforms from the Samson Stage 5 system for inputs at 100Hz, 5kHz and 15kHz respectively. It can be seen that the shape of the output is roughly constant regardless of the input frequency, although clipping is evident at the 100Hz output (this was found to be the most boosted frequency in Figure 9.1).

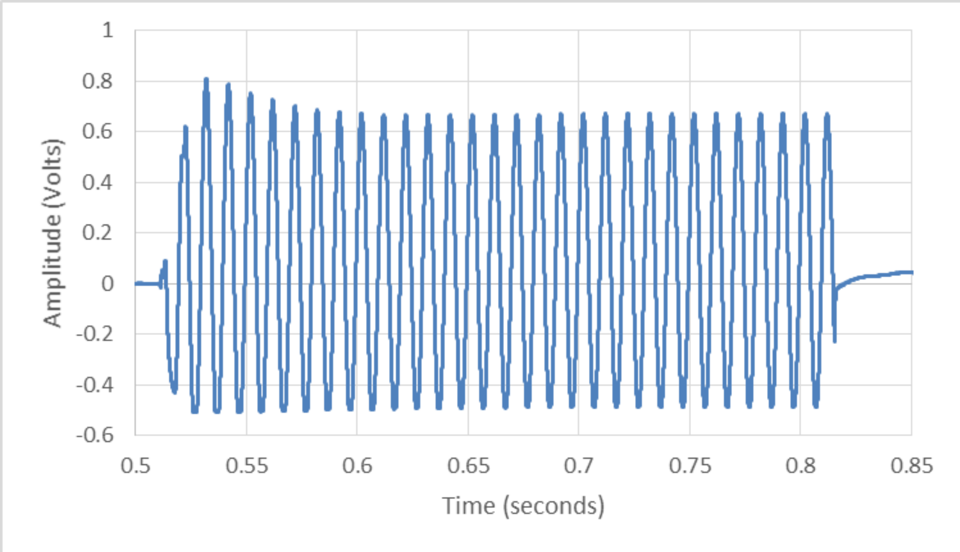

Figure 9.9 – Samson Stage 5 Response waveform to a burst of 100Hz sine waves

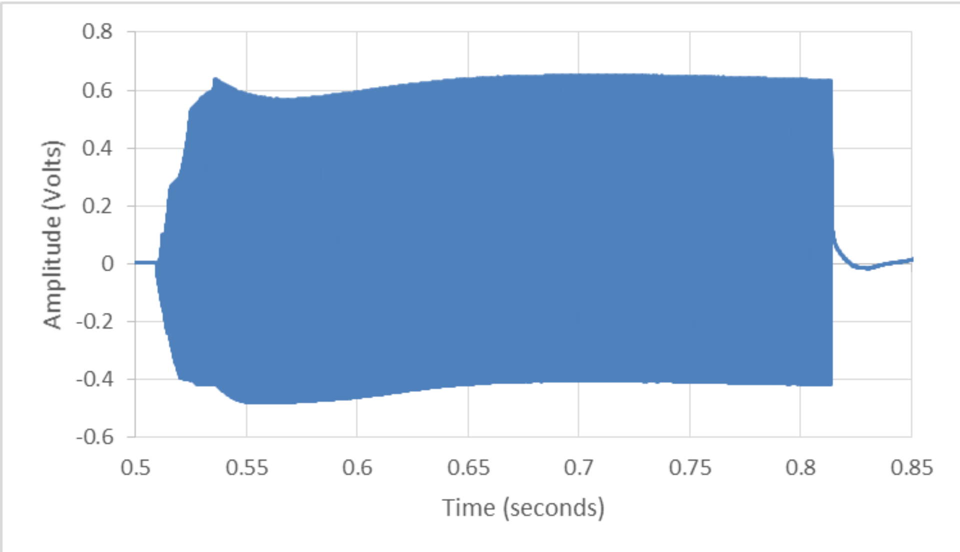

Figure 9.10 – Samson Stage 5 Response waveform to a burst of 5kHz sine waves

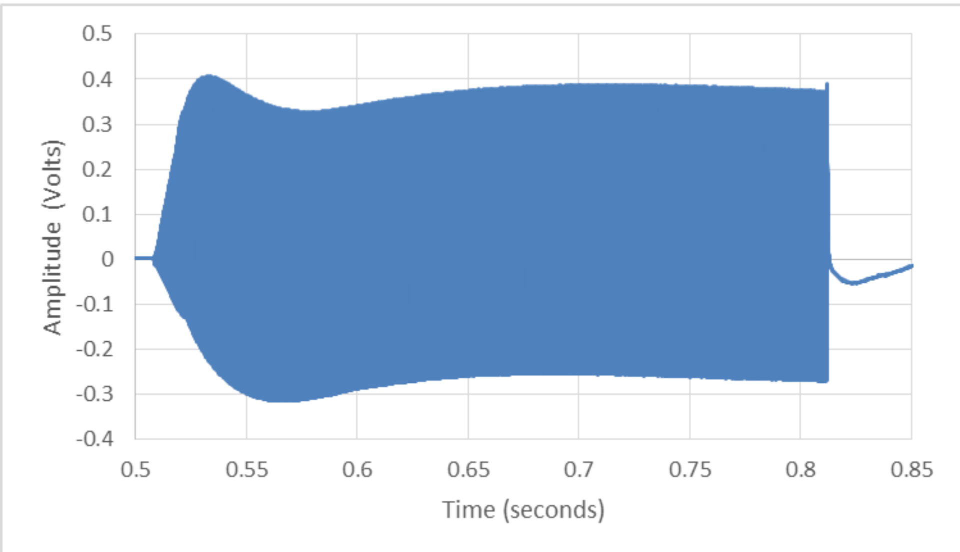

Figure 9.11 – Samson Stage 5 Response waveform to a burst of 15kHz sine waves

Steady state, in terms of peak-to-peak amplitude and ignoring any DC bias is reached at different points from the application of the input in each case, ranging from around 40ms for a 5kHz output to approximately 90ms for 100Hz.

9.1.5 Robustness

The frequency response curve as produced by the Frequency Sweep test conducted on the Samson Stage 5 system in the reverberation chamber with direct line-of-sight between transmitter and receiver is shown on Figure 9.12. Remember that here, the frequency axis may be viewed as time, increasing frequency representing increasing time within the adverse environment.

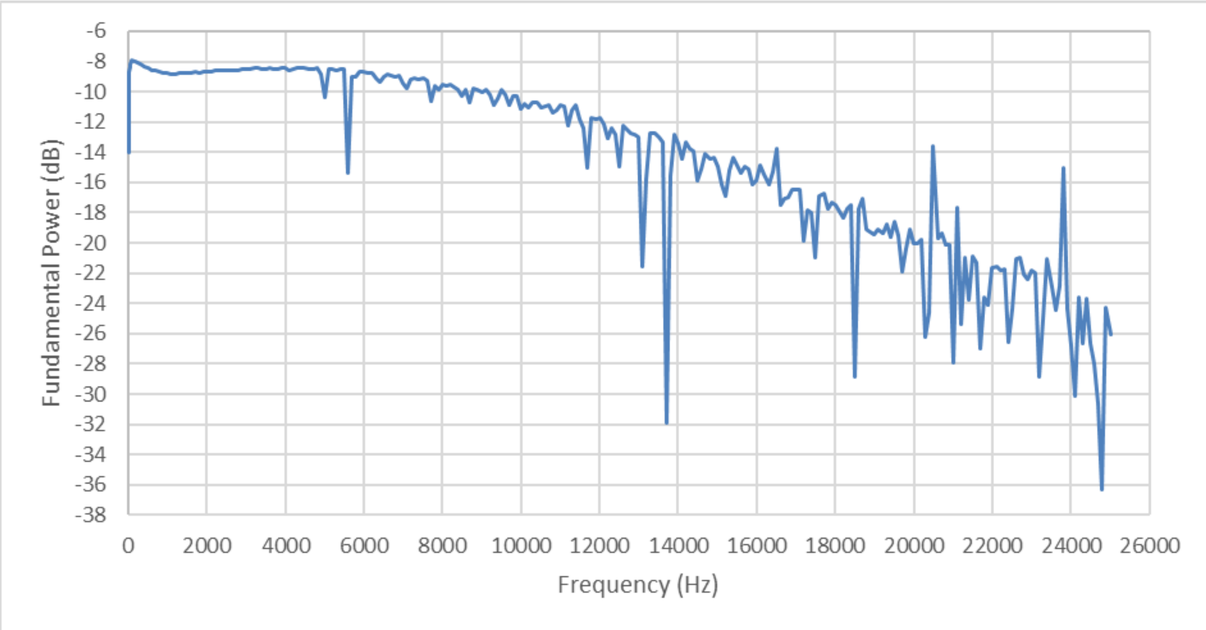

Figure 9.12 – Frequency Response curve for the Samson Stage 5 under the Line-Of-Sight test

Under the line-of-sight test, the effect of multipath interference on the Samson Stage 5 system is evident on Figure 9.12. Constructive and destructive interference leads to extreme points both above and below the expected power levels in the Frequency Response curve.

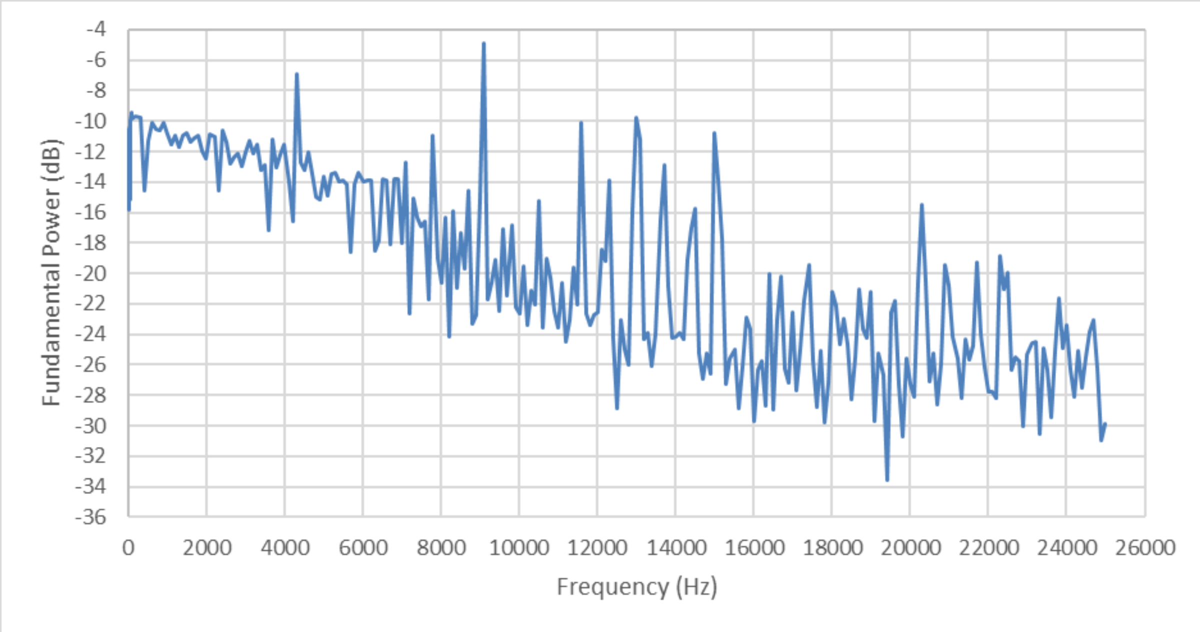

Figure 9.13 – Frequency Response curve for the Samson Stage 5 under the Non-Line-Of-Sight test

Removing the line of sight, the quality of the analogue wireless link further deteriorates. This is manifested as decreased signal purity at the receiver. The Frequency Response curve of Figure 9.13 shows a lot of fluctuations across the entire range, indicating problems throughout the entire duration of the test.

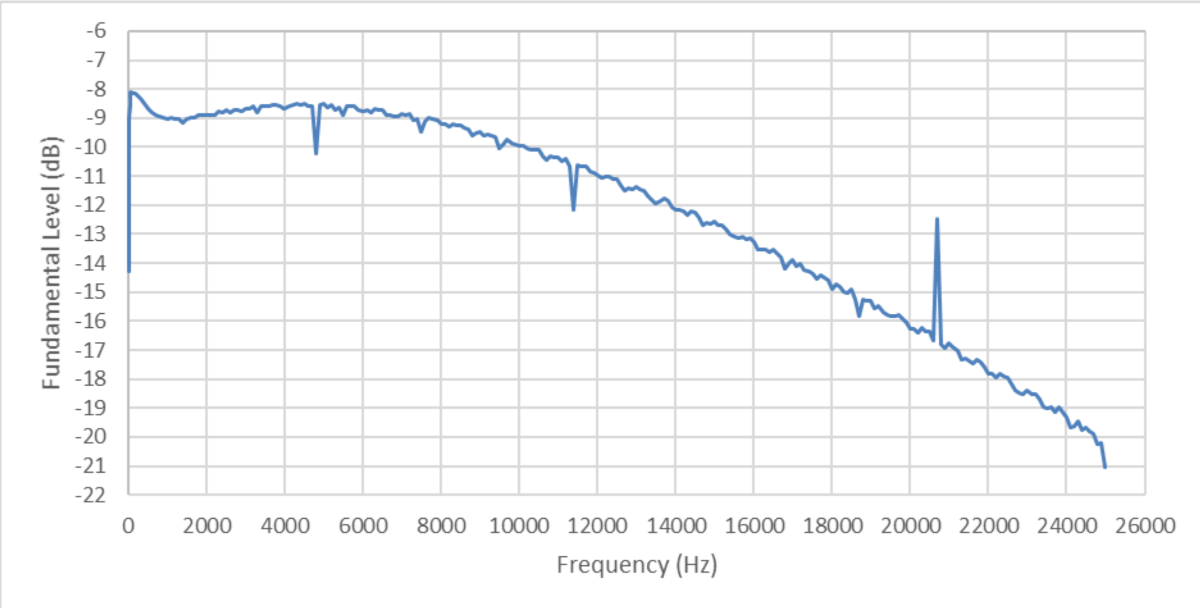

Figure 9.14 – Frequency Response curve for the Samson Stage 5 under the Phantom test

The presence of the Phantom device in the Reverberation Chamber improves the performance of the Samson Stage 5's wireless link. Comparing Figure 9.12; the frequency response of the system without the Phantom; to Figure 9.14; the response with the Phantom; it can be seen that sharp fluctuations in the curve are much less numerous and severe in the presence of the Phantom, indicating more stable performance throughout the duration of the test. This is due to the effect of the Phantom device of absorbing electromagnetic energy, reducing the level and abundance of multipath signals such that the line-of-sight signal has much greater power at the receiving antenna than the multipath interferers.

9.2 Joyo JW-01

9.2.1 Frequency Response

The frequency response of the Joyo JW-01 digital system is shown in Figure 9.15.

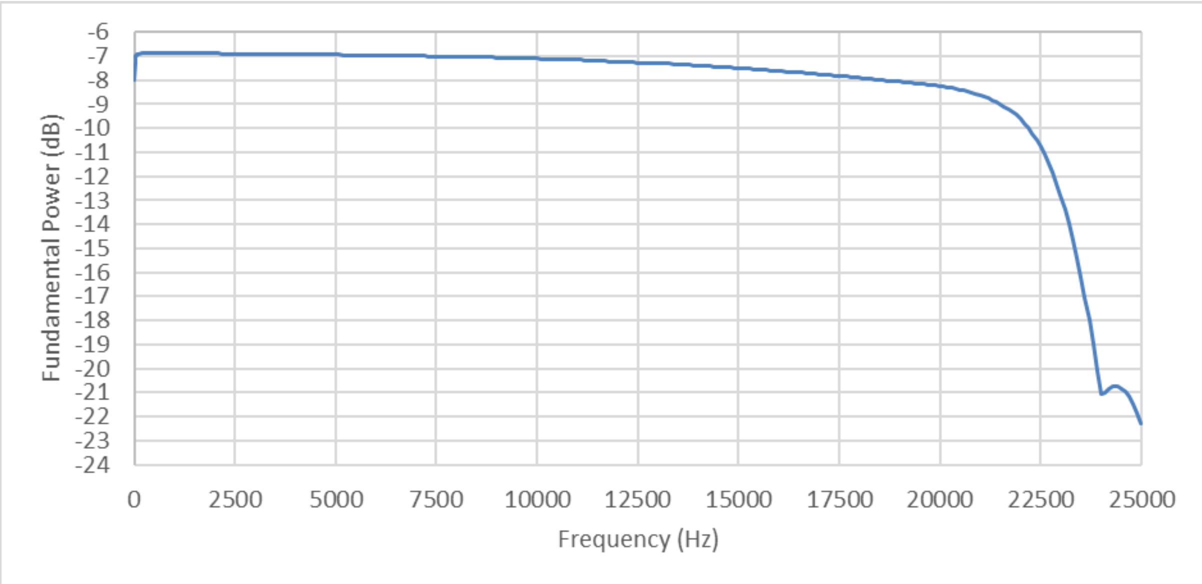

Figure 9.15 – Joyo JW-01 Frequency Response curve

The Joyo JW-01 exhibits a fairly flat frequency response in the 20Hz-20kHz band, peaking at -6.88dB at 1200Hz; a gain of 2.15dB (or 1.28 as a ratio) over the power of the input sine wave. Beyond this frequency, the response exhibits a gradually increasing roll-off gradient until about 21kHz. The -3dB point, with respect to the response at 1kHz, occurs between 22100Hz and 22200Hz; linear interpolation yields an approximate upper bound of 22151Hz. The Joyo JW-01 yields particularly impressive low-frequency response; -7.97dB at 10Hz, just -1.08dB from the level at 1kHz.

9.2.2 Harmonic Distortion

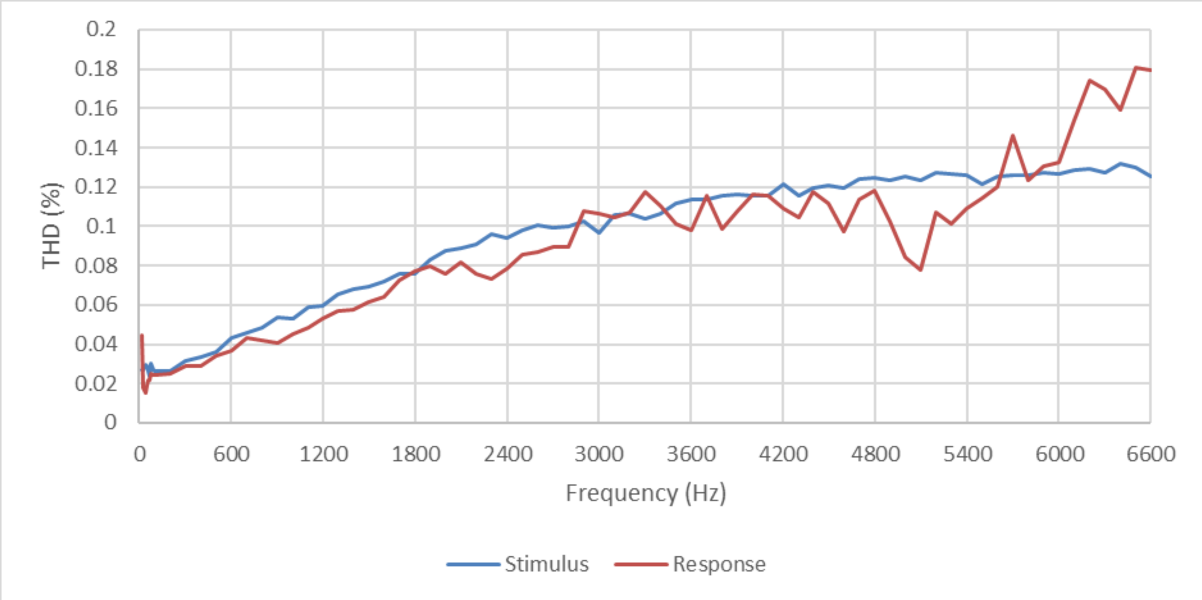

Figure 9.16 – THD vs Frequency for the Joyo JW-01, showing measurements for the response and stimulus signals

Figure 9.16 shows the Total Harmonic Distortion characteristic of the Joyo JW-01 system from 20Hz to 6.6kHz, as well as of the corresponding stimulus signal as produced by the ELVIS Function Generator. Looking at the THD measurement curves, it can be noted that the system is very insusceptible to harmonic distortion, so much so that the THD measurement for the response signal fluctuates either side of the curve for the stimulus signal. This is due to the random nature of noise, which can cause both additive and destructive interference to the frequency components of a signal. It can thus be concluded that the system adds virtually no harmonic distortion.

9.2.3 Signal Quality

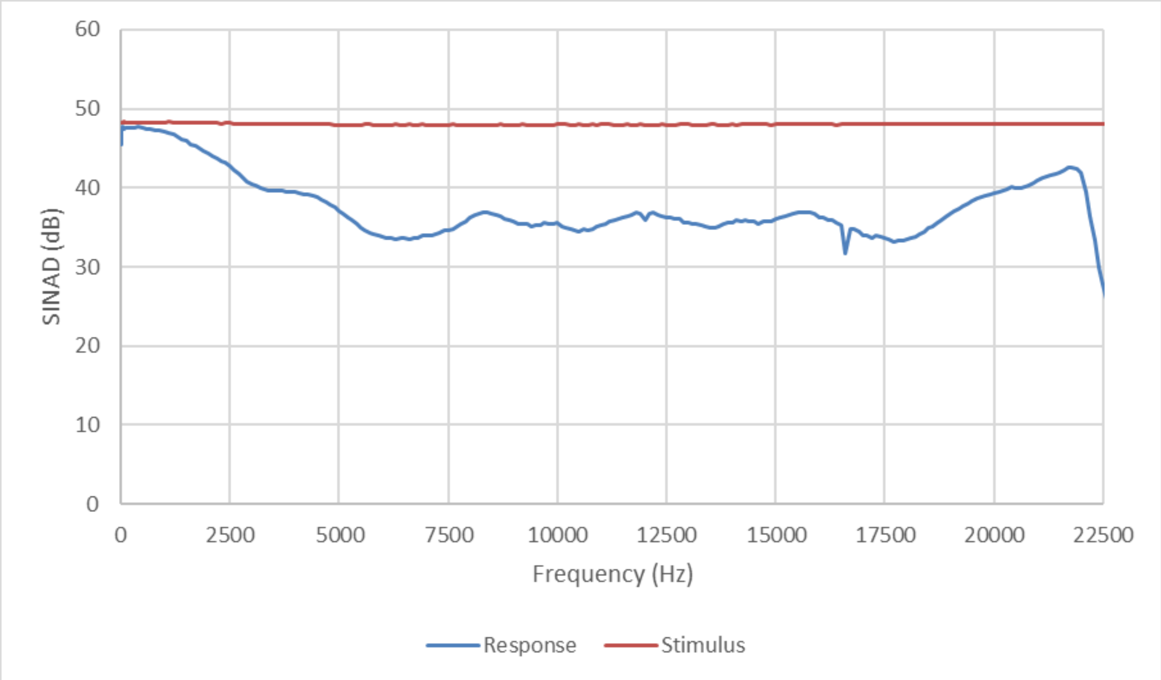

Figure 9.17 – SINAD vs Frequency for the Joyo JW-01, showing measurements for the response and stimulus signals

Figure 9.17 shows the SINAD characteristic of the Joyo JW-01 across the audible frequency band. For low input frequencies the delivered signal power, is found to be very high with respect to noise (and distortion), yielding SINAD measurements close to that for the input signal. However, as the frequency increases the SINAD decreases to below 37dB, approximately 11dB below the measurement for the input signal, within the range 5kHz to 19kHz. Since Figure 9.16 shows the system to be insusceptible to harmonic distortion, this indicates increased noise output across this range. Increasing signal quality with respect to distortion and noise at the upper end of the system's frequency response indicates that the system outputs less noise as the output fundamental level decreases.

The fluctuations in SINAD measurement across the audible range indicate that the noise output from the Joyo JW-01 does not remain at a constant power but is dynamic according to the input frequency. Figure 9.18 further demonstrates this point by showing the power spectra of the outputs for 1kHz and 6kHz inputs.

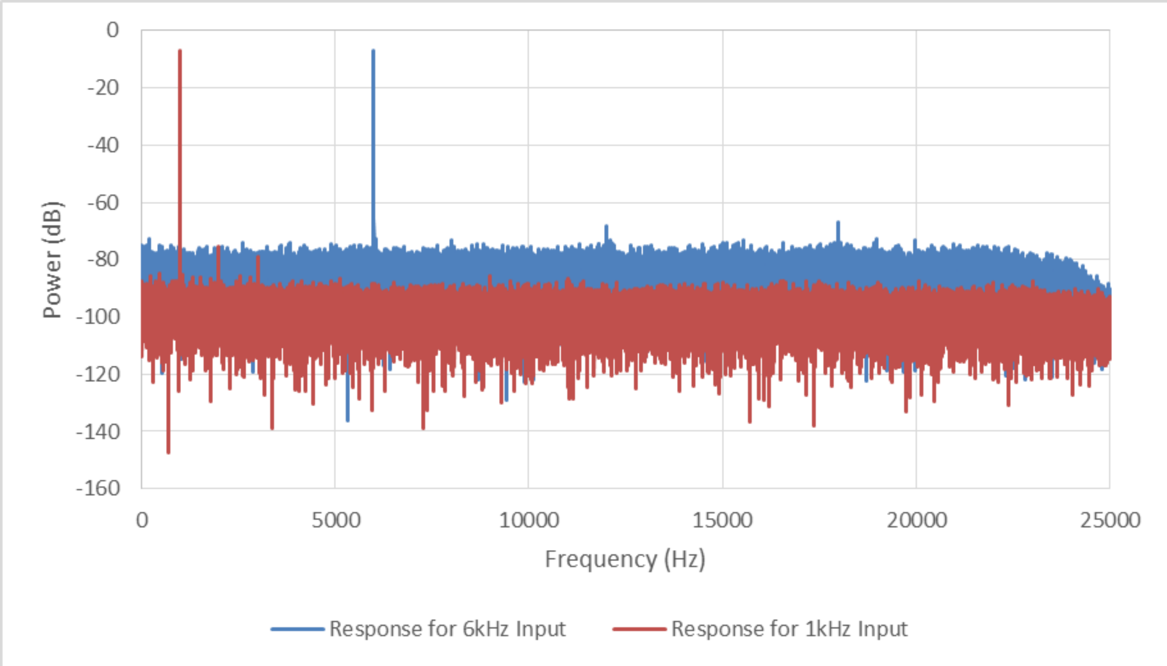

Figure 9.18 – Power spectra for the Joyo JW-01 showing the frequency compositions of the responses for input frequencies of 6kHz and 1kHz

It can be seen from Figure 9.18 that the noise output is increased across the range by approximately 13dB for a 6kHz input signal than for a 1kHz signal. This accounts for the 13.24dB decrease in SINAD measurement from the 1kHz input to the same for the 6kHz input; from 47.07dB SINAD to 33.83dB.

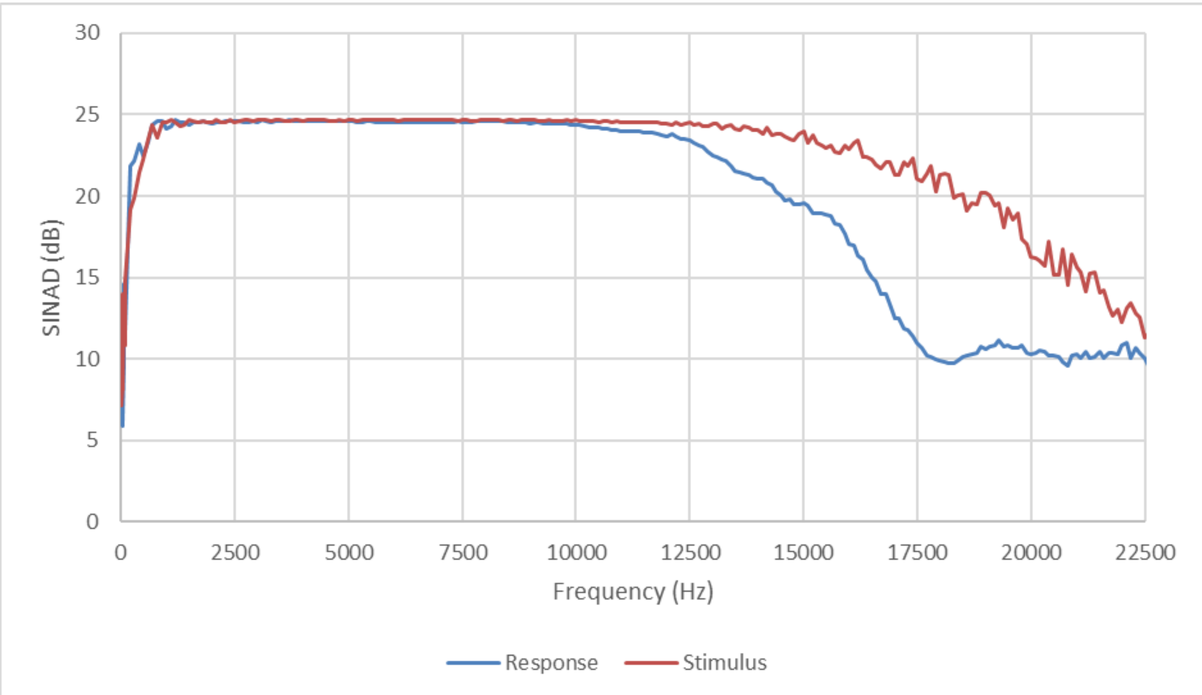

Figure 9.19 - SINAD vs Frequency for the Joyo JW-01, showing both stimulus and response measurements, under ITU-R 468 Noise Weighting

Filtering the Joyo JW-01's output according to the MATLAB-designed ITU-R 468 weighting filter, the system yields similar SINAD measurements to the input signal for frequencies lower than 10kHz. This indicates that within this range the noise power produced at the output of the receiver is evenly distributed across the range, i.e. white noise. Beyond 10kHz, the SINAD measurements fall to a maximum of 11.78dB below the stimulus signal at 17.8kHz, indicating the abundance of noise power at lower frequencies more detectable by the human ear, with respect to the filtered fundamental power.

9.2.4 Latency

The entire burst of input sine waves and corresponding output from the Joyo JW-01 at 1kHz is shown in Figure 9.20 below.

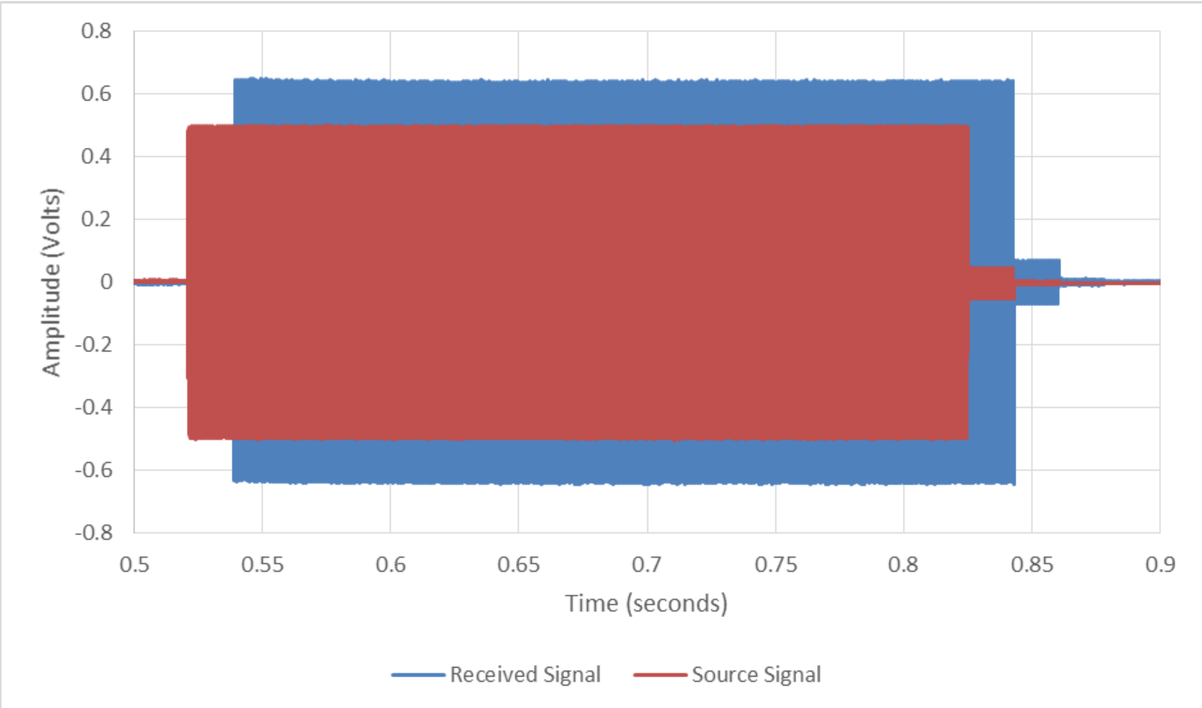

Figure 9.20 - Stimulus and response waveforms for the Joyo JW-01 under the latency test using a 1kHz frequency

Zooming in on these waveforms, the initial few cycles of both the input and output signals can be viewed in higher resolution, as shown on Figure 9.21.

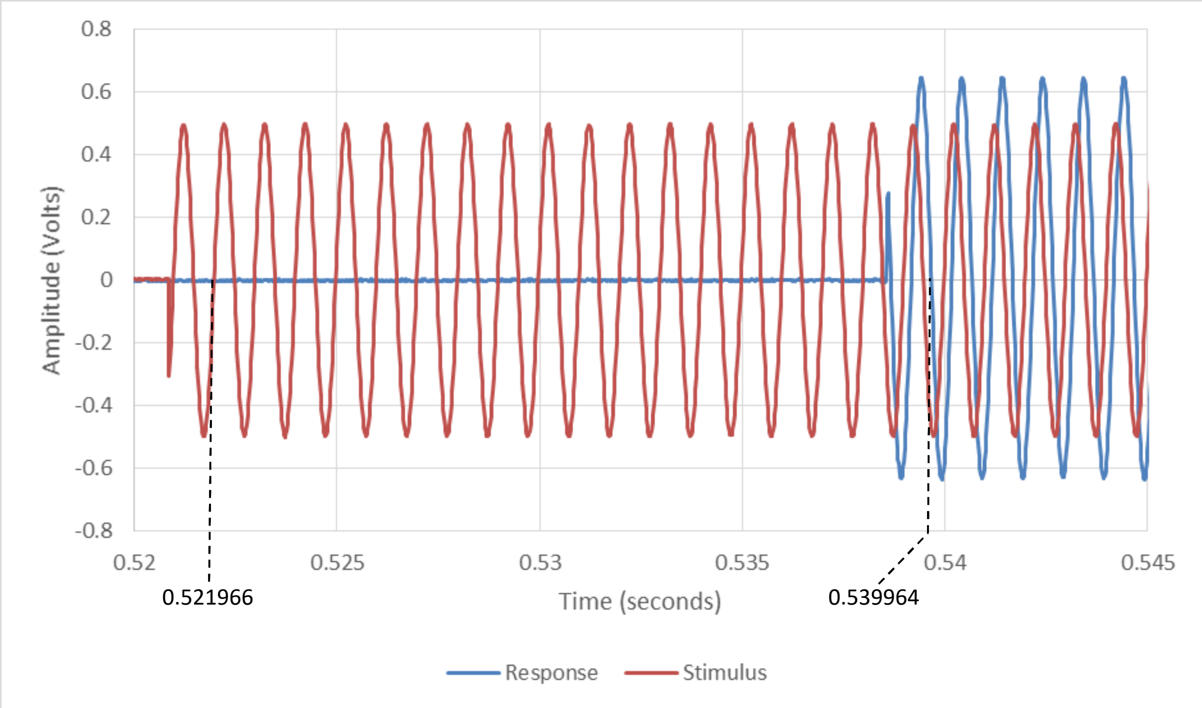

Figure 9.21 – Initial cycles of the stimulus and response waveforms for the Joyo JW-01 under the latency test using a 1kHz frequency

Visible on Figure 9.21 are the Latency Test waveform responses from the beginning of source signal generation to its appearance at the output of the receiver. As indicated on the figure the first full zero-to-zero cycle of the generated 1kHz signal ends at 0.521966s and the corresponding point for the received signal occurs at 0.539964s (interpolated zero-intercepts). The latency introduced by the Joyo JW-01 can thus be quantified as the time difference between these two points, at 0.017998s or about 18ms.

Quantifying the latency in the same method for input fundamental frequencies of 100Hz, 5kHz, 10kHz and 15kHz yields measurements of 17.39ms, 17.65ms, 18.03ms and 17.34ms respectively. This demonstrates the latency introduced by the system to be independent of the input frequency components, at approximately 17.7ms ± 0.4ms.

Interestingly, the receiver output signal appears to have opposite polarity to the source signal, i.e. a positive input voltage yields a negative output voltage (and vice-versa). This is evident on Figure 9.21 as the initial negative-voltage ‘spike’ seen in the source signal is induces a corresponding positive spike in the receive signal. Not counting the latency, this results in the output signal being 180° out of phase with respect to the input signal. However, since human hearing is insensitive to changes in phase, this will have no impact on the perceived sound.

9.2.5 Robustness

The Joyo JW-01 digital wireless system proved to be very susceptible to multipath interference, such that it was rendered completely unable to operate within the Reverberation Chamber under line-of-sight, non-line-of-sight and Phantom conditions. However, it was observable that the system was able to operate as normal when the door of the chamber was left wide open, greatly reducing the effect of multipath links within the chamber by allowing RF signals to escape.

In order to characterise and demonstrate the response of this system, then, to adverse wireless conditions, a 1kHz signal was applied to the input of the transmitter unit and the corresponding output signal captured with the chamber door opened by varying amounts. With the chamber door completely closed, the received signal SINAD ratio was calculated at just 1.59dB, while the stimulus signal read 48.35dB. This indicates that the system was completely unable to recognise the signal transmitted by the transmitter unit under heavy multipath interference, outputting only a low-power noise signal. Opening the door as wide as possible yielded a SINAD ratio of 47.02dB, as the system was able to operate as normal. Half-opening the chamber door, the system suffered many dropouts and erroneous packets, evident from the noisy, intermittent output signal observable in Figure 9.22. The SINAD measurement in this case was understandably low, at 4.73dB.

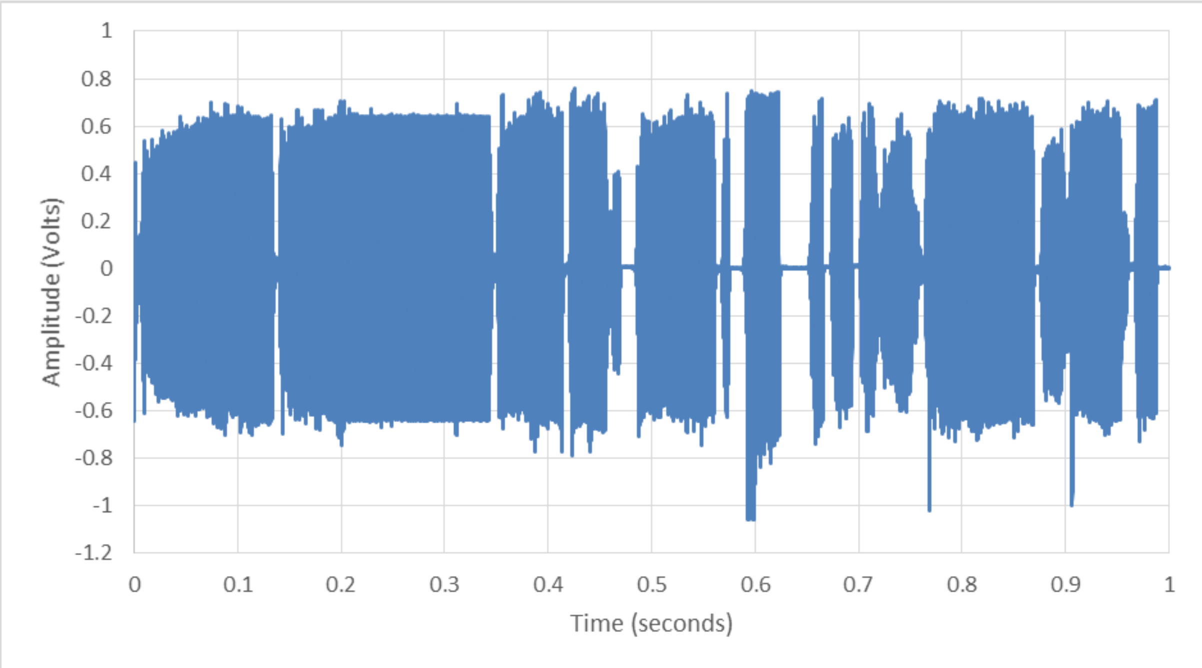

Figure 9.22 – Joyo JW-01 output waveform for a 1kHz input in the reverberation chamber with the door half-open

9.3 Laird DVK-BTM511 (SBC Codec)

9.3.1 Frequency Response

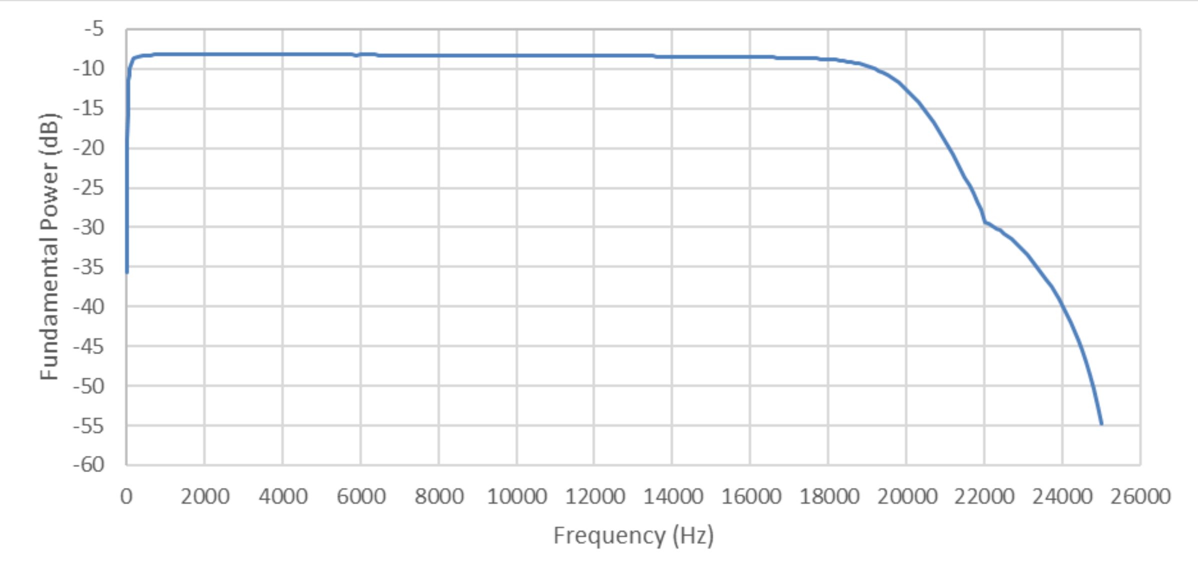

Figure 9.23 – Laird DVK-BTM511 Bluetooth (SBC Codec) Frequency Response curve

The frequency response of the Laird DVK-BTM511 Bluetooth setup is shown in Figure 9.23. Similarly to the other digital system under test; the Joyo JW-01; the system exhibits a mainly flat frequency response across the majority of the audible range. The gain at 1kHz with respect to the -9.03dB input signal is 0.83dB, a ratio of 1.10. The lower and upper -3dB points relative to 1kHz for this system lie at 73.57Hz and 19642Hz respectively.

9.3.2 Harmonic Distortion

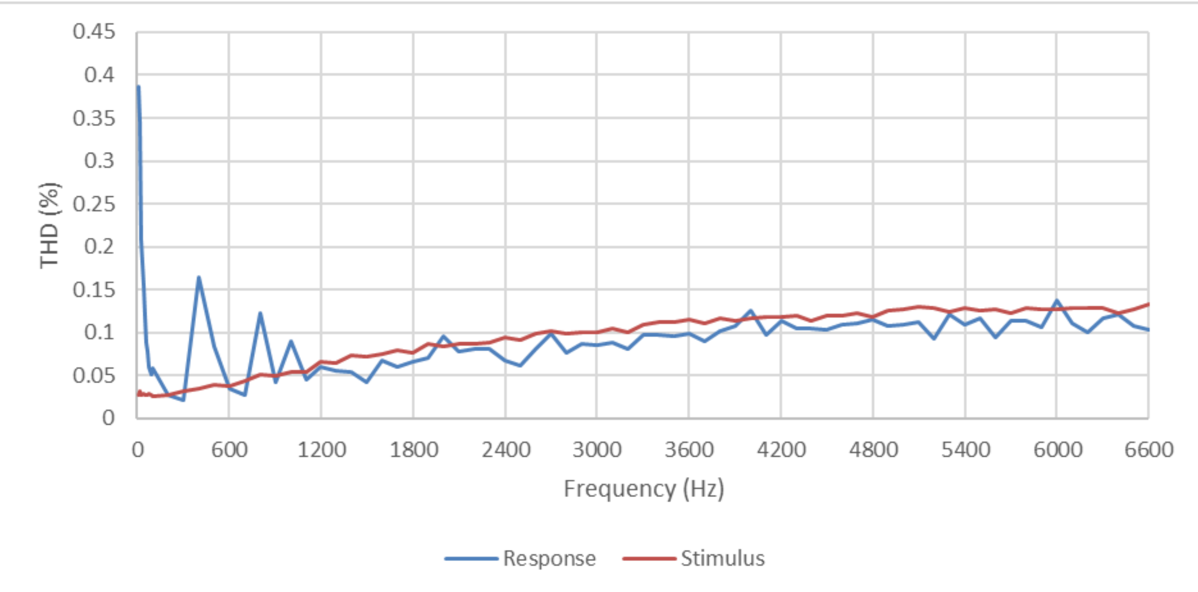

Figure 9.24 - THD vs Frequency for the Laird DVK-BTM511 Bluetooth (SBC Codec) system, showing measurements for the response and stimulus signals

It can be seen from the Total Harmonic Distortion characteristic of the Laird DVK-BTM511 Bluetooth system (Figure 9.24) that the harmonic content of the audio signal is essentially unaltered by the system across most of the tested frequency range. The fluctuations in the response characteristic at the lower end of the spectrum are most likely to be due to additive noise at the harmonic frequencies, rather than actual variations in the tendency of the system to cause harmonic distortion. Also, the relatively high proportion of harmonic distortion at the extreme low end, i.e. below 100Hz, is due to the fact that the signal gain is low for these frequencies, according to the system's Frequency Response (see Figure 9.23), such that the noise level is high in relation to the signal level.

9.3.3 Signal Quality

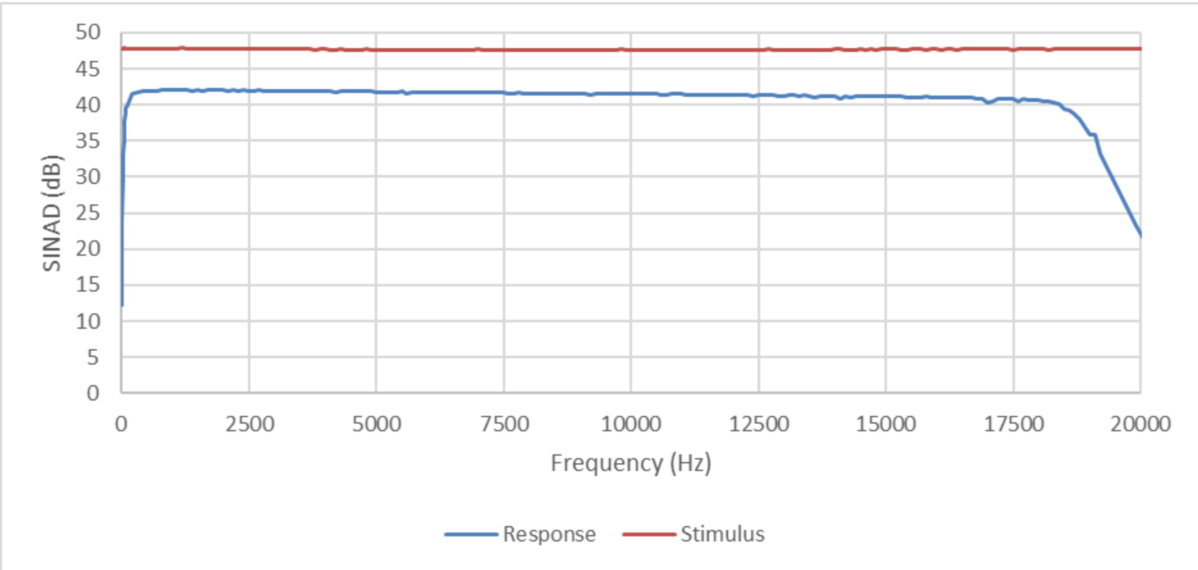

Figure 9.25 - SINAD vs Frequency for the Laird DVK-BTM511 Bluetooth (SBC Codec) system, showing measurements for the response and stimulus signals

Figure 9.25 shows the SINAD characteristic for the Laird DVK-BTM511 Bluetooth system across its frequency response range. The measurement remains fairly constant throughout much of the range, falling steadily from 42dB at 1.7kHz to 40dB at 18.4kHz, with the input signal at approximately 48dB SINAD. This correlates well with the Frequency Response graph of Figure 9.23, indicating that noise power remains mostly constant with respect to the signal power, regardless of the its fundamental frequency. As expected, SINAD falls at the outer extremes of the frequency range as the fundamental power also falls.

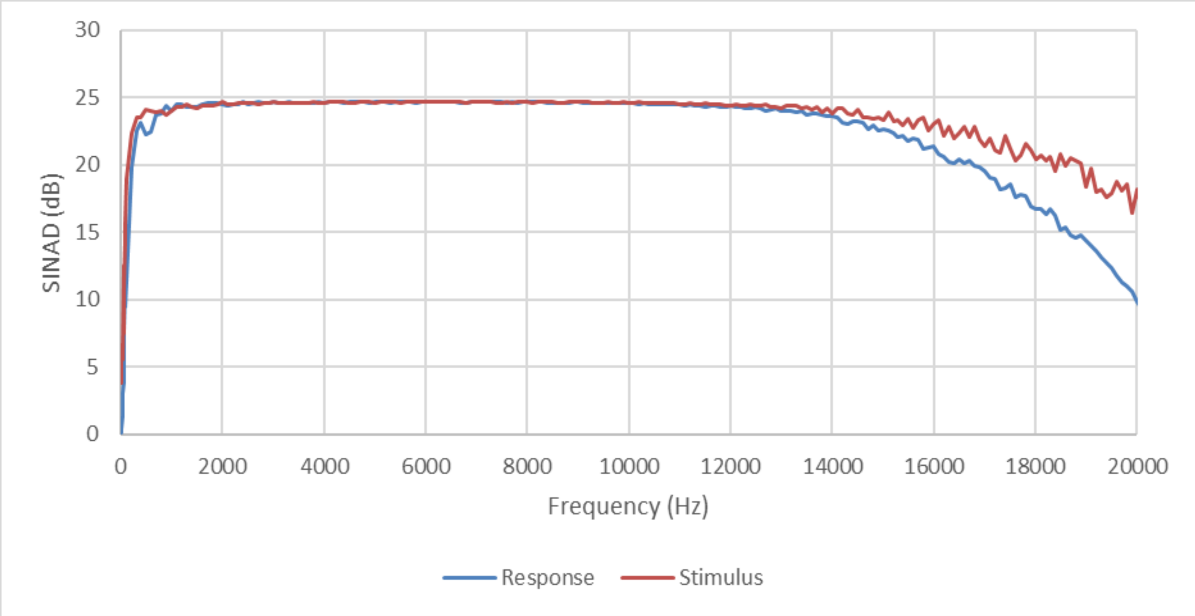

Figure 9.26 - SINAD vs Frequency for the Laird DVK-BTM511 Bluetooth (SBC Codec) system, showing measurements for the response and stimulus signals, under ITU-R 468 Noise Weighting

Figure 9.26, the SINAD characteristic of the Laird DVK-BTM511 Bluetooth system using the SBC codec under ITU-R 468 Noise Weighting, indicates that the noise output remains mostly constant across the frequency range, as does the stimulus signal. As the frequency response drops off towards the top of the audible range, so too does the SINAD measurement for the weighted response signal.

9.3.4 Latency

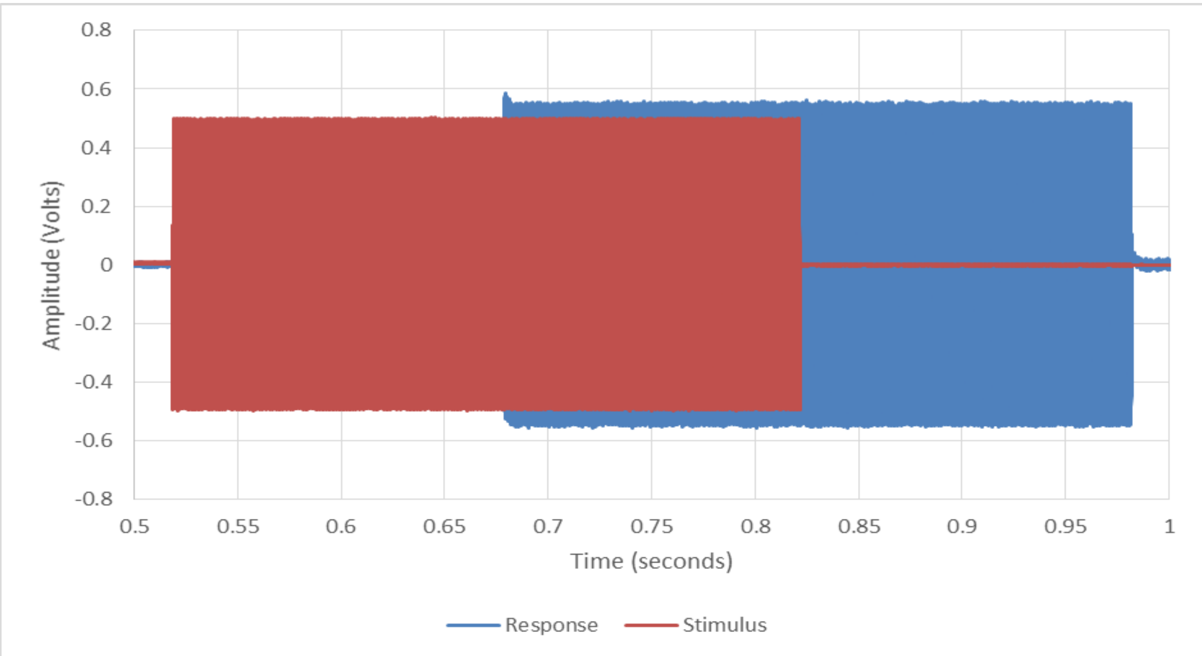

Figure 9.27 - Stimulus and response waveforms for the Laird DVK-BTM511 Bluetooth (SBC Codec) system under the latency test using a 1kHz frequency

The Laird DVK-BTM511 Bluetooth development kit, running A2DP audio streaming using the standard SBC codec, has significant latency in comparison with the previous systems, as can be seen from the entire response under the Latency Test in Figure 9.27. As before, measuring the latency as the time difference between the end of the first full zero-to-zero sine wave of the source signal to the corresponding point in the received wave, yields a value of 0.159918s, or 159.92ms.

The points on the input and output waveforms from which the latency measurement was made for an input tone of 1kHz are clearly shown on Figures 9.28 and 9.29 respectively.

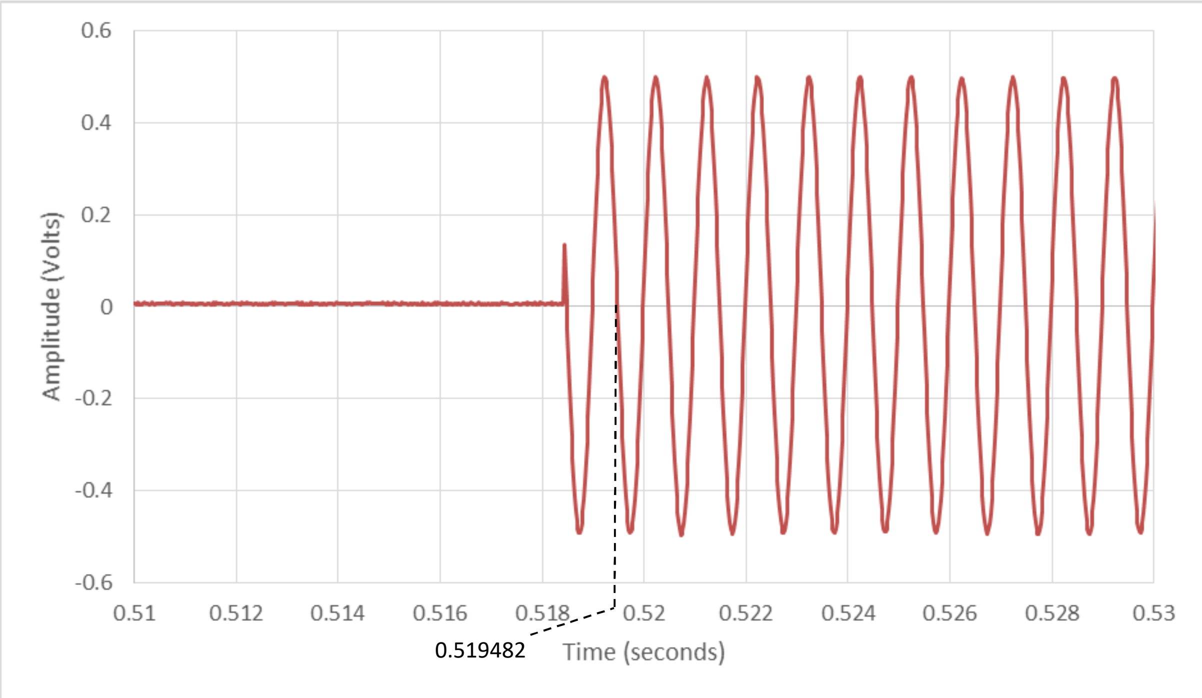

Figure 9.28 – Stimulus waveform for the Laird DVK-BTM511 (SBC codec) latency test using a 1kHz frequency, showing the point from which latency was measured

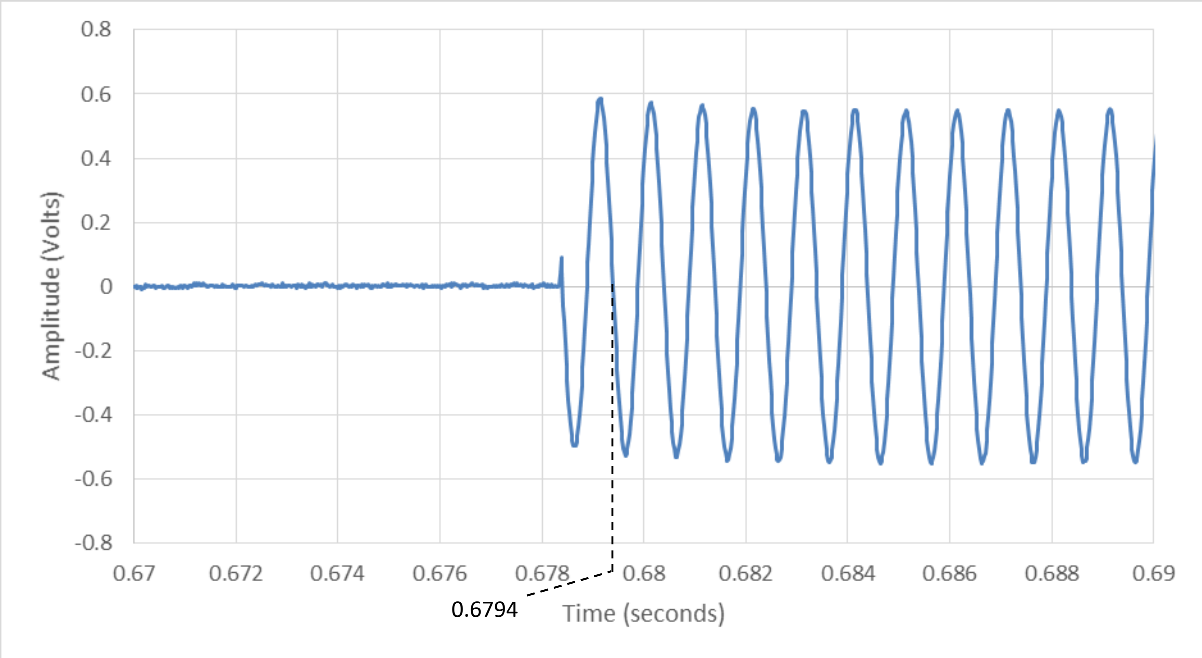

Figure 9.29 – Response waveform for the Laird DVK-BTM511 (SBC codec) latency test using a 1kHz frequency, showing the end point for latency measurment

For input frequencies at 100Hz, 5kHz, 10kHz and 15kHz the measured latency values are 158.05ms, 159.58ms, 159.66ms and 159.82ms respectively. Unlike the Joyo JW-01, the latency of the Bluetooth system under the SBC codec appears to increase slightly with increasing input frequency. This may be due to algorithmic differentiations between separate frequency bands. The measured latency for this system can be specified at 159ms ± 1ms.

9.3.5 Robustness

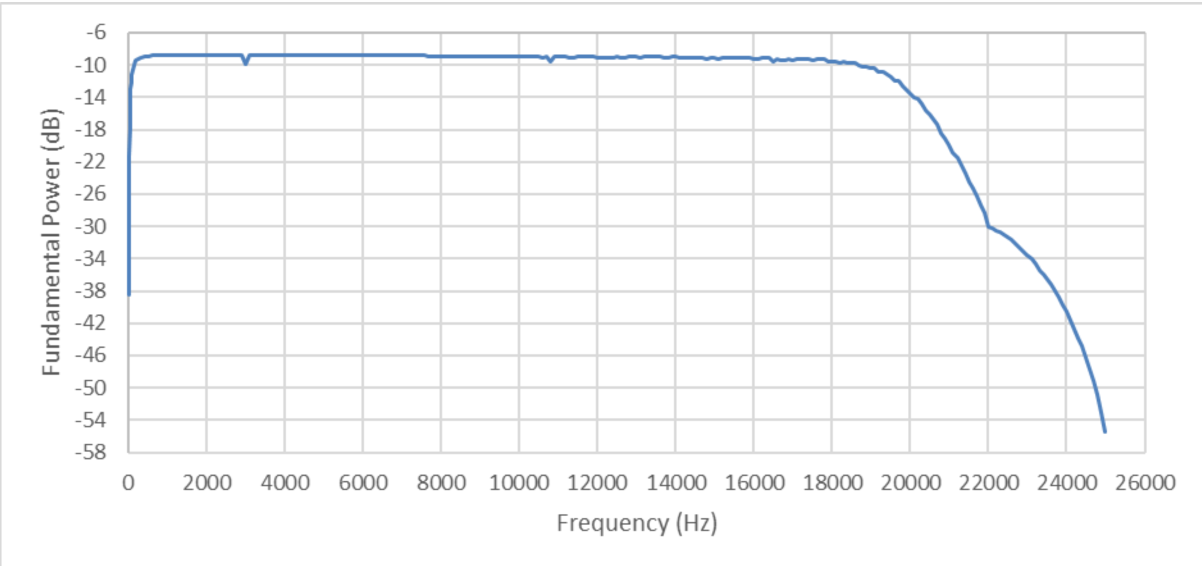

Figure 9.30 – Frequency Response curve for the Laird DVK-BTM511 Bluetooth system under the Line-Of-Sight test

The Laird DVK-BTM511 Bluetooth system appears more resilient to multipath interference in comparison with the Samson Stage 5 analogue system, in that its Frequency Response curve under the Line Of Sight Reverberation Chamber test is much smoother and 'cleaner' than the same for the analogue system, with much fewer and less severe fluctuations from the expected trend. Conclusively the Bluetooth link yields a more reliable transmission link under these conditions.

The improvement in performance of the Bluetooth system over the Joyo JW-01 system is largely due to Bluetooth's frequency hopping behaviour. Since the channel used for transmission is switched for each packet, the receiver becomes insensitive to interference at frequencies other than those within the current channel. However the system does still exhibit reduced performance compared to typical conditions, as multipath conditions results in the presence of residual RF noise across the 2.4GHz spectrum leading to increased bit error rate.

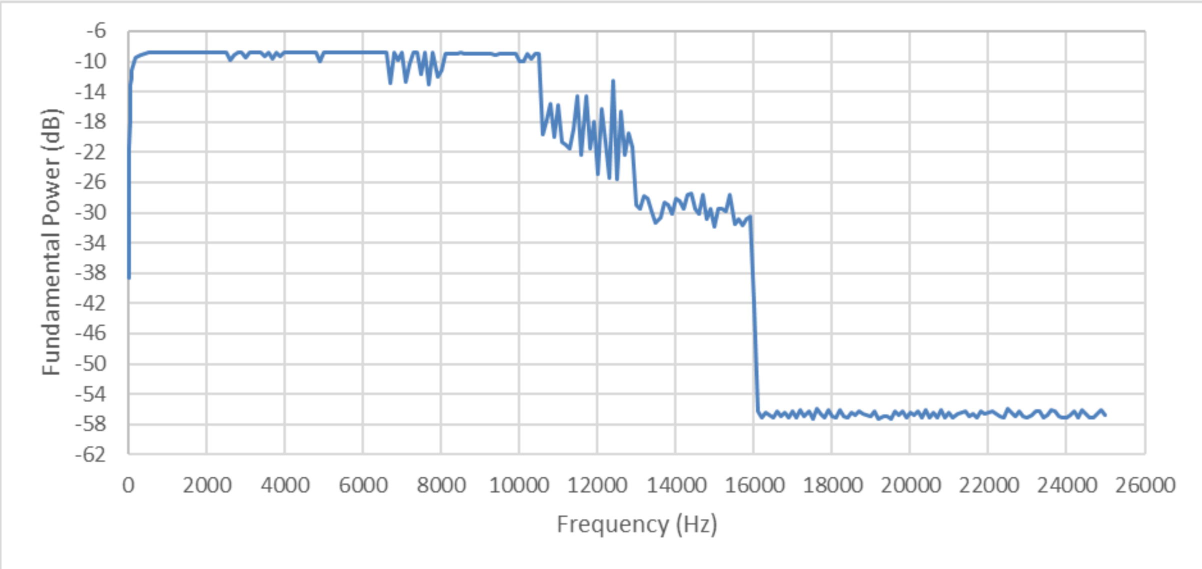

Figure 9.31 - Frequency Response curve for the Laird DVK-BTM511 Bluetooth system under the Non-Line-Of-Sight test

Removing the line of sight connection between transmitter and receiver, the Bluetooth link suffers serious connection problems. As evident from the Frequency Response curve of Figure 9.31, under the non-line-of-sight test, following a fairly steady initial connection the link suffered an increasing number of drop-outs, eventually losing communications altogether. Once the system has lost connection, the Bluetooth link must be manually re-initiated before communications are restored. This allows resynchronisation of the FHSS scheme and other parameters which will be lost by the receiving module once the initial link is broken.

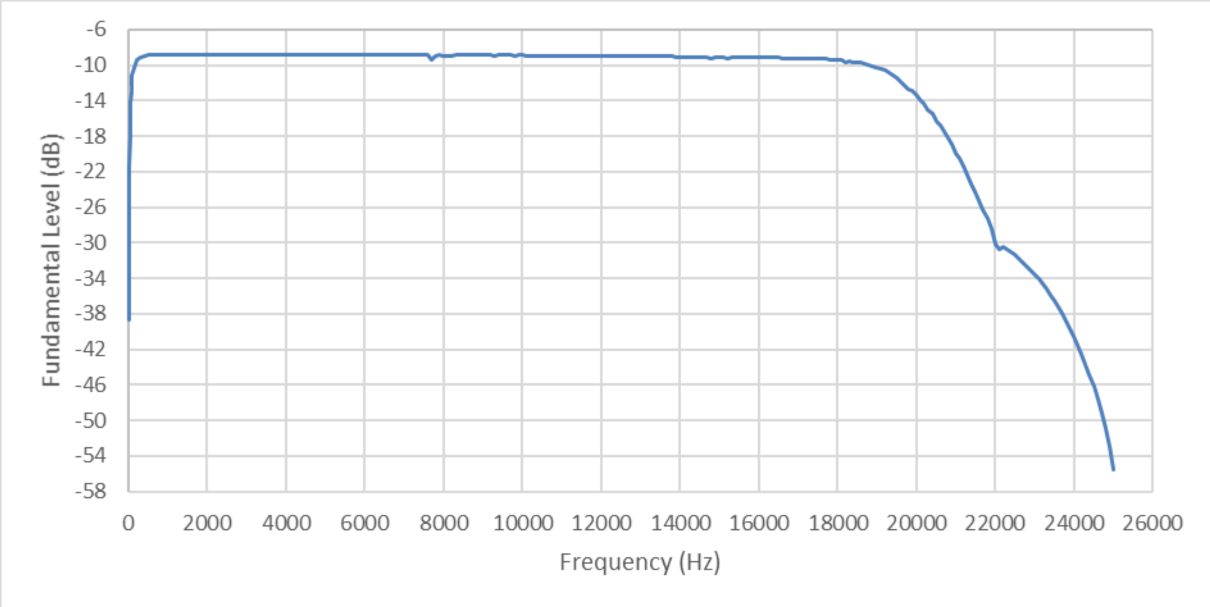

Figure 9.32 - Frequency Response curve for the Laird DVK-BTM511 Bluetooth system under the Phantom test

The introduction of the Phantom device to the Reverb Chamber test has had no observable effect on the operation of the Bluetooth link, as demonstrated by Figure 9.32. This is expected since Bluetooth is already resilient to interference.

Note that it is not expected that the Bluetooth system would exhibit any change in performance in the adverse environments under application of the aptX codec. Since the codec layer deals only with data compression and decoding algorithms it will not have any effect on the robustness of the Bluetooth link. As such the behaviour exhibited by the Bluetooth system is assumed independent of codec employed.

9.4 Laird DVK-BTM511 (aptX Codec)

9.4.1 Frequency Response

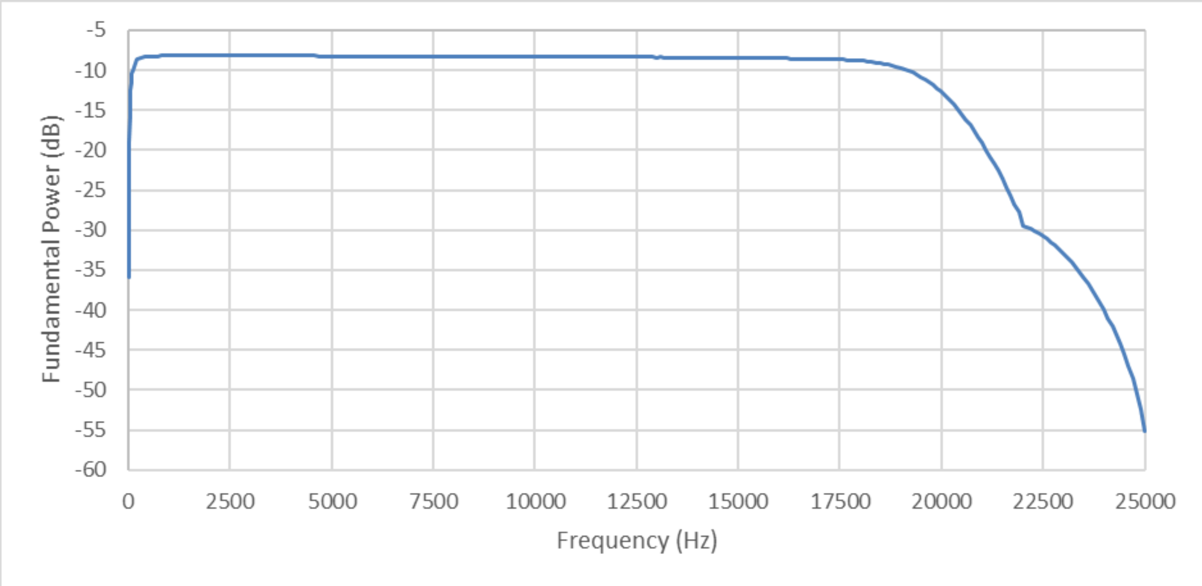

Figure 9.33 – Laird DVK-BTM511 Bluetooth (aptX codec) Frequency Response curve

Interestingly, the use of the aptX codec seems to have had very little impact on the frequency response of the Bluetooth setup, comparing Figure 9.33 with Figure 9.23. The power level at 1kHz remains the same as under the SBC codec, at -8.2dB, representing a gain of 0.83dB or 1.10 as a ratio. Under the aptX-enabled system the interpolated -3dB points relative to 1kHz lie at the lower end of the audible spectrum at 74.48Hz and at the upper end at 19624Hz.

9.4.2 Harmonic Distortion

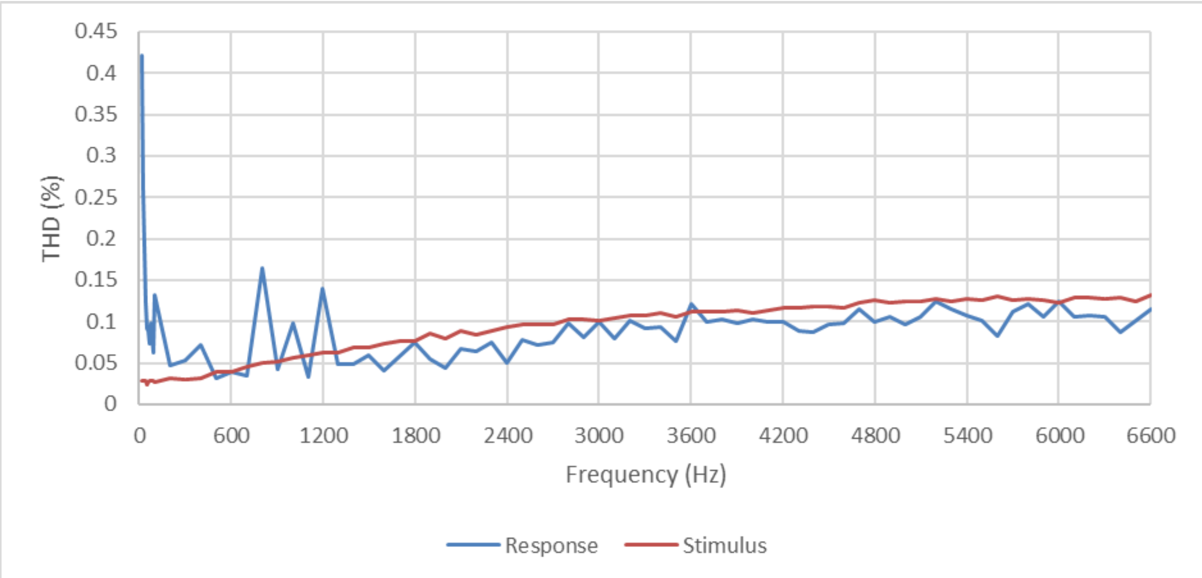

Figure 9.34 - THD vs Frequency for the Laird DVK-BTM511 Bluetooth (aptX codec) system, showing measurements for the response and stimulus signals

Similarly to the results for the SBC codec and the Joyo JW-01, the Laird DVK-BTM511 has virtually no effect on the audio signal in terms of harmonic distortion, as shown on the Total Harmonic Distortion characteristic curve of Figure 9.34. The characteristic has similar artefacts to those discussed under section 9.3.2, and so it can be concluded that the use of the aptX codec has had no effect on the harmonic distortion characteristic of the Bluetooth system.

9.4.3 Signal Quality

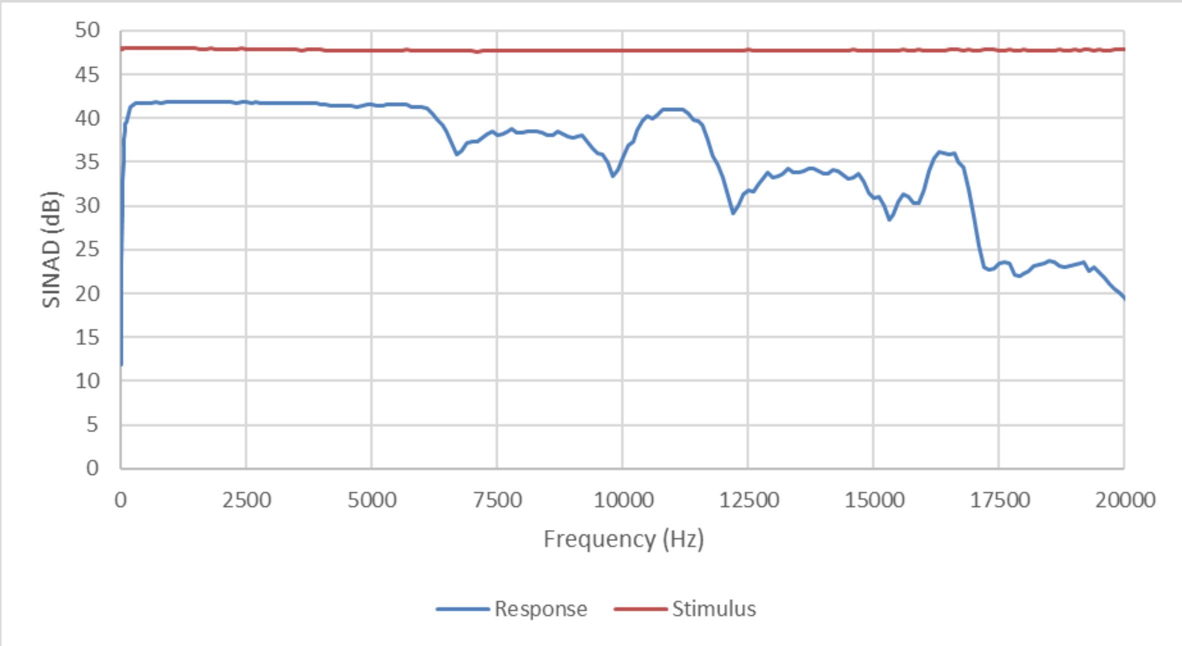

Figure 9.35 - SINAD vs Frequency for the Laird DVK-BTM511 Bluetooth (SBC Codec) system, showing measurements for the response and stimulus signals

The SINAD characteristic produced by the Laird DVK-BTM511 Bluetooth system reveals some interesting behaviour of the aptX codec. As shown in Figure 9.35, the SINAD characteristic follows the same trend as for the SBC codec (see Figure 9.25) for the low frequencies up to about 6.5kHz. Beyond this, the spectrum is divided into a number of bands for which the SINAD curve follows distinct patterns with generally decreased signal quality.

Analysing the spectral composition of the output signals within two of these bands further light is shed on the behaviour of the aptX codec. Figure 9.36 below shows the power spectra for the output signal at inputs of 18kHz and 9.5kHz.

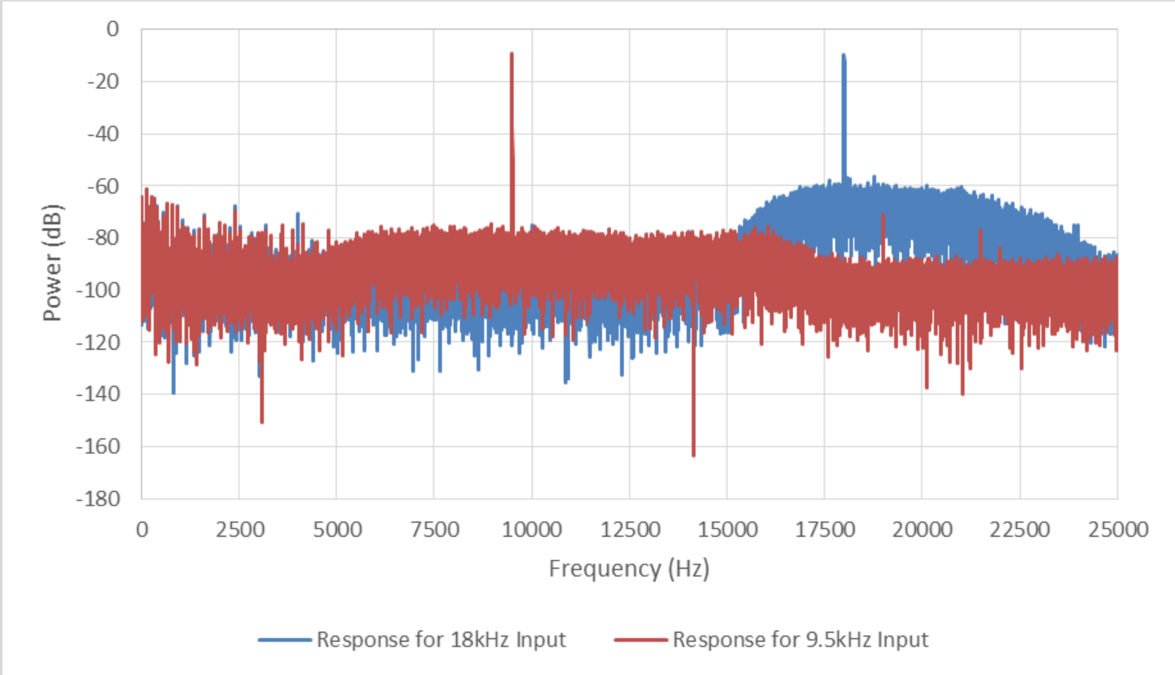

Figure 9.36 - Power spectra for the Laird DVK-BTM511 system (aptX codec) showing the frequency compositions of the responses for input frequencies of 18kHz and 9.5kHz

As evident from Figure 9.36, the aptX codec has the effect of boosting frequency components within a certain range, dependent on the frequency of the input signal. Since the inputs applied under the Frequency Sweep test are single tones, the boosting the frequencies within a certain range has the effect of raising a section of the noise floor, thereby reducing the signal quality as measured by the SINAD metric. This behaviour of the aptX codec is further discussed under Chapter 10 *Discussion*.

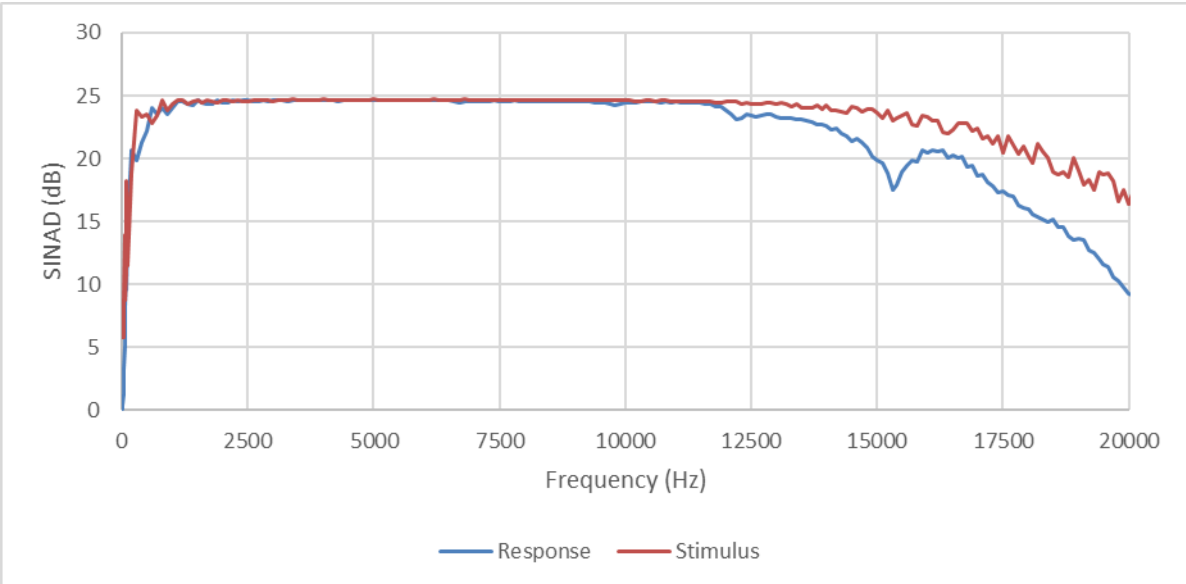

Figure 9.37 - SINAD vs Frequency for the Laird DVK-BTM511 Bluetooth (SBC Codec) system, showing measurements for the response and stimulus signals under ITU-R 468 Noise Weighting

As for the previous systems, the effect of Noise Weighting on the SINAD measurement characteristic across the range is to alter the shape of the response graph of Figure 9.37 to follow that of the stimulus signal across the lower section of the spectrum. As signal quality and fundamental response power decrease towards the upper bound of the audible range, the SINAD measurement for the weighted signals also decreases.

9.4.4 Latency

It can be seen from Figure 9.38 that the use of the aptX codec has not majorly improved upon the latency of the Bluetooth system.

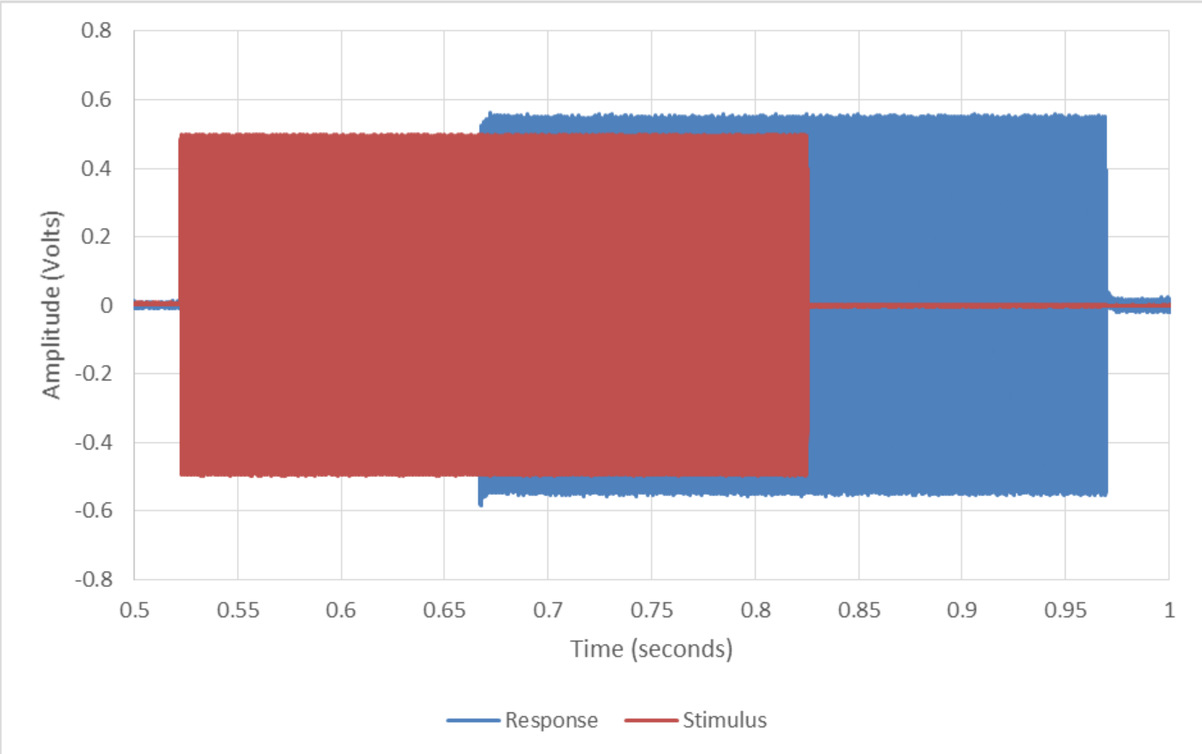

Figure 9.38 - Stimulus and response waveforms for the Laird DVK-BTM511 Bluetooth (aptX Codec) system under the latency test using a 1kHz frequency

Figures 9.39 and 9.40 show in greater detail the initial few cycles of the stimulus and response waveforms respectively, including the interpolated time values for the zero-crossings following the first full sine waves. Computing the difference between these values yields a latency measurement of 0.148135s, or 148.14ms.

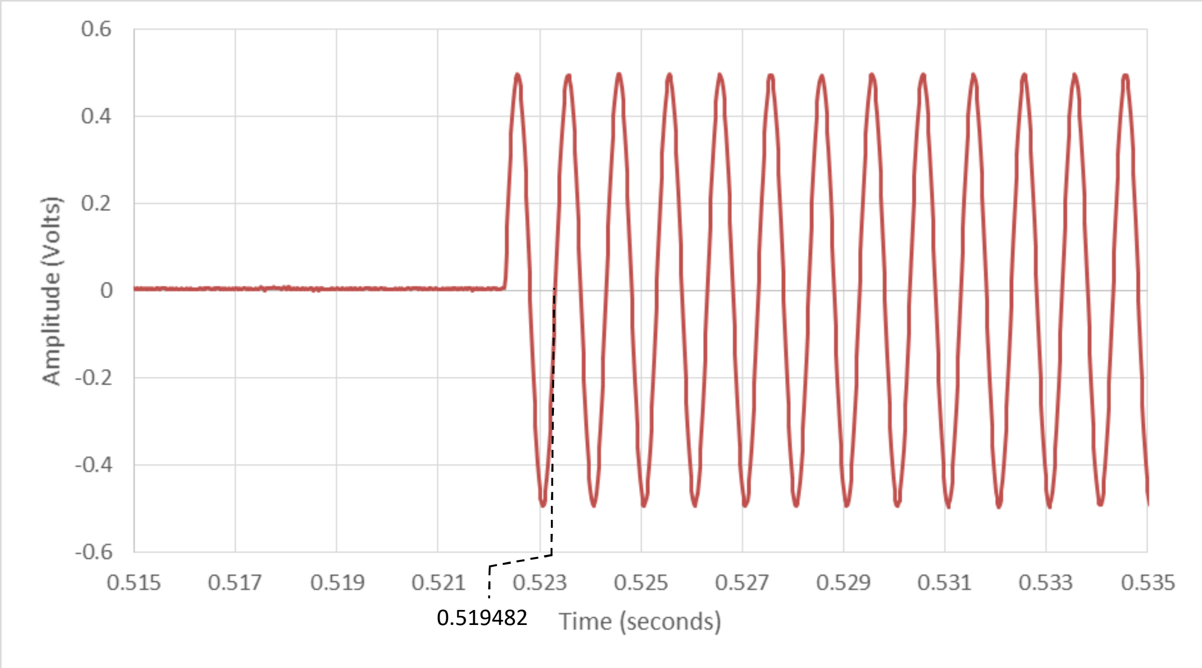

Figure 9.39 - Stimulus waveform for the Laird DVK-BTM511 (aptX codec) latency test using a 1kHz frequency, showing the point from which latency was measured

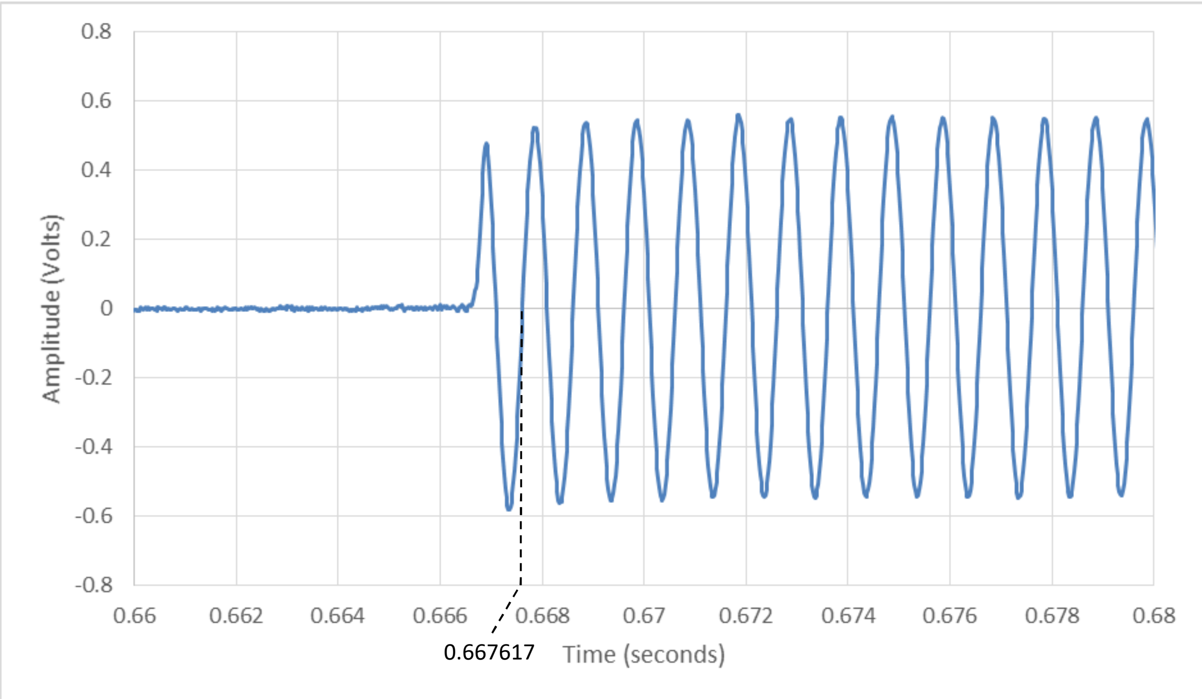

Figure 9.40 - Response waveform for the Laird DVK-BTM511 (aptX codec) latency test using a 1kHz frequency, showing the end point for latency measurment

Inputting signals at 100Hz, 5kHz, 10kHz and 15kHz as before, the latency of the aptX-enabled Bluetooth system is measured at 147.07ms, 148.65ms, 148.71ms and 148.82ms respectively. As with SBC, the latency appears to increase slightly with increasing frequency, indicating possible algorithmic differences within the codec for different frequency bands. The tests conducted on the system yield an approximate latency measurement of 148ms ± 1ms, an overall improvement on SBC of about 11ms due to improved algorithmic efficiency.

10. Discussion

10.1 Experimental Conclusions

Compiling the results of all the tests conducted on each of the wireless audio transmission systems it is possible to deduce a number of conclusions regarding their operation considering delivered audio quality, latency and robustness.

10.1.1 Audio Quality

The Samson Stage 5 analogue system's frequency response of 15Hz to 13485Hz is the narrowest of all those tested, as well as being the most variable with frequency within its passband. This system was also most susceptible to harmonic distortion, second and third harmonic components reaching up to 1.2% of the energy in the fundamental in the 1.3kHz to 6.6kHz range. The tests have also demonstrated the signal quality, measured as the SINAD ratio, to be variable across the frequency range of the Samson Stage 5 system; being lowest at approximately 11dB below the input SINAD within the same 1.3kHz to 6.6kHz band that sees most harmonic distortion.

The Joyo JW-01 system manifested the widest frequency response of all those tested, from below 10Hz to over 22151Hz. While unsusceptible to harmonic distortion, the system did however demonstrate a dynamic noise floor with respect to input frequency. Low frequencies below about 100Hz yielded impressive SINAD ratios of just 1dB below the input SINAD, but increasing the input fundamental to 5kHz and above, greater noise output reduced SINAD measurements to a minimum of 14.6dB below the input SINAD. Interestingly the system's frequency response was fairly non-flat in comparison to the Bluetooth system as discussed later exhibiting a shallow roll-off between 10kHz and 20kHz. This may be the result of DSP algorithms to simulate the low-pass filter characteristics of traditional long-length coaxial instrument cables.

Under both SBC and aptX codecs, Bluetooth audio streaming technology, as represented by the Laird DVK-BTM511 development kit setup, yields a flat frequency response over much of the audible range, falling just short at the upper and lower frequency bounds at approximately 75Hz and 19.6kHz respectively. Like the Joyo JW-01 digital system, Bluetooth proved not to affect the harmonic composition of the input signal, while SINAD measurements revealed some interesting differentiations between the proprietary 2.4GHz protocol and Bluetooth, and between A2DP audio codecs operating over Bluetooth. The SBC codec demonstrated relatively constant signal quality across much of the audio range, SINAD measurements remaining between 6dB and 8dB below the input SINAD between 100Hz and 18.4kHz. For the aptX codec, however, the measured signal quality appeared to vary with frequency across the range. As demonstrated earlier in the report, the aptX codec has the effect of boosting certain frequency bands at the output depending on the fundamental frequency of the input signal. For the single tone inputs employed under the frequency sweep test, this resulted in increased noise output at frequencies close to the fundamental, causing decreased SINAD measurements. While difficult to objectively measure, such equalisation may yield higher perceived sound quality for music transmission, the application for which the aptX codec is primarily designed.

10.1.2 Latency

The nature of analogue modulation and demodulation allows for extremely low latency in terms of signal generation at the output following the application of an input signal, but as the tests have shown the initial signal quality can much inferior to the input, being distorted and low-amplitude. The effect of this analogue filter behaviour to the listener is a change in the 'attack' of the signal, i.e. the speed at which it reaches steady amplitude from zero. Latency measurements on the Samson Stage 5 revealed the settling time of the system, i.e. the time taken to reach steady-state output amplitude, to vary from approximately 40ms to 90ms. As outlined in Chapter 5 *Audio Quality Assessment*, research conducted by the Audio Engineering Society found 42ms to be the maximum appropriate latency for wedge monitoring situations, such that the settling time characteristic manifested by the Samson Stage 5 is likely to lie outside the undetectable limit.

Overall, the Joyo JW-01 exhibited the most impressive latency measurements at approximately 17.7ms. While greater than the 10ms delay claimed by the manufacturer, this is still well below the maximum allowable limit as suggested by the AES. Like the Bluetooth implementations the latency tests showed the initial output after application of an input signal to be much higher quality than that for the analogue system, better retaining the attack of the input signal.

Bluetooth demonstrated much greater delays than the other tested systems under both the SBC and aptX codecs. While aptX was able to reduce the latency by about 11ms to approximately 147ms from 158ms for SBC due to increased algorithmic efficiency, the Bluetooth delay remained well above the 42ms maximum acceptable limit for real-time musical instrument monitoring.

10.1.3 Robustness

As expected the analogue technology employed by the Samson Stage 5 system proved very susceptible to multipath interference under the non-ideal conditions tests. Under the Line-Of-Sight and particularly the Non-Line-Of-Sight tests, the system's performance was shown to be much worsened in the presence of multipath interference. Performance was however improved under the Phantom test, indicating the effect of human tissue of electromagnetic radiation absorption.

Comparing Bluetooth to the Joyo JW-01's proprietary digital communications protocol, it is evident that the latter has had to sacrifice robustness in order to achieve its low latency, ease of pairing and features such as one-to-many operation. Regarding ease of pairing, the Joyo system requires only a one-off connection process, when presumably the transmitter and receiver establish which wireless channel to operate on, allowing multiple receivers to listen to the same channel for one-to-many operation. Conversely, Bluetooth requires increased handshaking to establish the FHSS sequence, profile, encryption parameters etc that will be used for communication, that are not persisted through a power cycle. This leads to increased handshaking delay and difficulties reconnecting when the link is broken. While completely unable operate under any of the reverberation chamber tests indicating high susceptibility to interference, the Joyo JW-01 was able resume communications as soon as conditions were restored. Conversely, once the Bluetooth link was broken mid-way through the Non-Line-Of-Sight test, it required manual reconnection. The extra overhead required by Bluetooth to enable advanced communications techniques does however greatly improve its robustness as demonstrated by the Laird DVK-BTM511 system's comparatively good performance under multipath interference.

10.2 Issues Regarding Frequency-Domain Analysis

Many of the measurements and characterisations made of audio systems require the application of the Fourier Transform to map time-domain signals into the frequency domain. The FFT (Fast Fourier Transform) algorithm employed by DSP platforms to perform this mapping assumes that the input time-domain signal is infinitely periodic, that is, the input signal repeats itself exactly over and over. This assumption is problematic because it cannot be ensured that the time-window sampled input signal contains an integer number of all frequency components, thus if the captured waveform is assumed to repeat, some erroneous pattern will occur at the point where the end of the wave is 'stitched' together with the beginning of the window. This phenomenon leads to spectral leakage in the frequency domain, where the energy contained in a specific component appears spread across a range of frequencies either side of the actual frequency. Clearly, this can lead to inaccurate frequency-domain measurements.

To counteract the problem of spectral leakage, a window function can be applied to the captured window of samples. Many window functions exist and have varying effects on the behaviour of the Fourier Transform, but they generally have the effect of dampening the signal in some way at the beginning and end of the sampling window, thereby reducing the power of leakage components. One common window function and the one employed by the Analysis VIs throughout this project is the Hanning window, whose effect in both the time- and frequency-domains are illustrated in Figures 10.1 and 10.2. Note that applying no window function is also known as rectangular windowing.

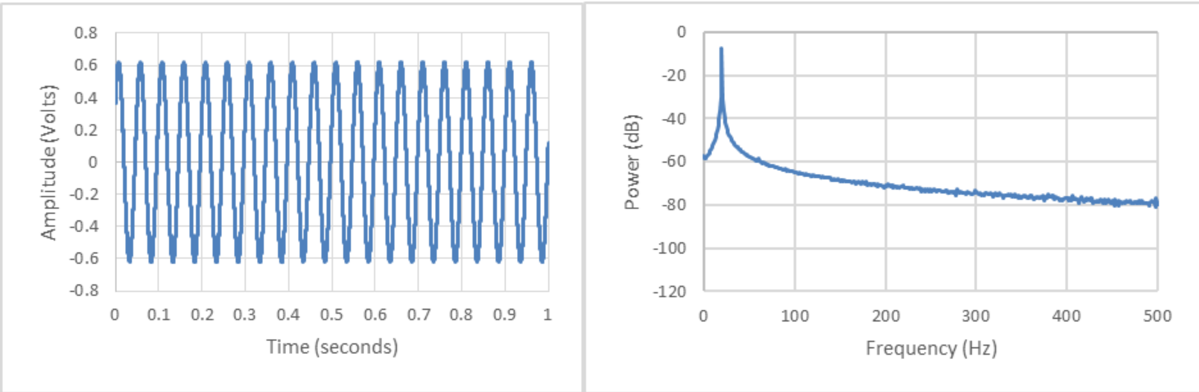

Figure 10.1 – Time domain signal and corresponding Frequency domain spectrum using a rectangular window for the Joyo JW-01's 20Hz response

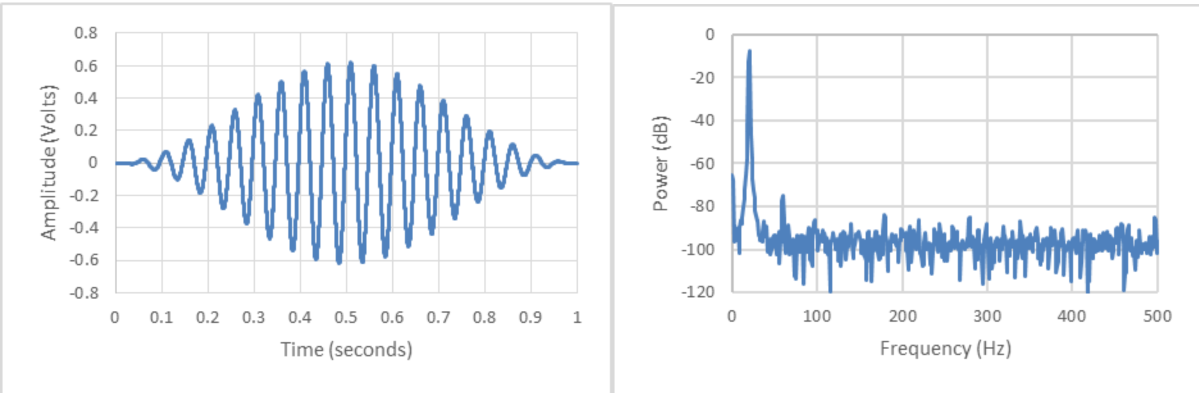

Figure 10.2 – Time domain signal and corresponding Frequency domain spectrum using a Hanning window for the Joyo JW-01's 20Hz response

It can be seen that the application of the Hanning window has the effect in the frequency domain of reducing spectral leakage, better exposing the individual frequency components contained within the input signal. This allows for more accurate characterisation of the frequency-domain makeup of the systems' responses.

Another problem with frequency-domain analysis on finite length signals is that even if the input is zero-average, there may be an imbalance in the number of positive-voltage samples taken with respect to negative-voltage samples. The assumption of continually periodic signals yields a DC bias artefact, which shows up in the frequency domain power spectrum as a spike at zero hertz. This is visible on power spectra throughout this report, including those in Figures 10.1 and 10.2 above.

It is worth noting that due to the impact of window functions on the results yielded by the Fourier transfer, measurements relating to harmonic distortion and frequency response are somewhat dependent on the function employed. The Hanning window function was found to yield largely similar performance in terms of noise levels and spectral leakage to other functions such as Blackman, Blackman-Harris and Hamming windows, while others, particularly the Dolph-Chebyshev window, yielded dramatically altered results. Accordingly, the Hanning function was selected as an appropriate window for all frequency-domain computations for the audio measurement application.

10.3 Limitations of Experimental Data

While the NI ELVIS platform provides a wide range of useful instruments and data acquisition interfaces, it is primarily an education and prototyping platform; it is not specifically designed for accurate high-fidelity audio equipment measurements. As such a number of properties of the ELVIS hardware were found to be non-ideal for the audio system characterisation application. As demonstrated under Chapter 9 *Test Results and Analysis* of this report, sine wave signals produced by the Function Generator output were impure, containing a proportion of harmonic distortion and noise. In fact, the ELVIS Function Generator has a relatively low resolution of 8 bits, meaning that it sweeps across 256 possible discrete voltage levels to produce pseudo-sinusoidal AC waveforms. Applying the Signal-to-Quantisation-Noise Ratio equation (equation 4.1), the maximum Signal-to-Noise Ratio of the Function Generator output is calculated as follows:

$$SNR \approx 6.02(8) + 1.76 = 49.98dB$$

Thus the ELVIS Function Generator has a theoretical maximum Signal-to-Noise Ratio of 49.98dB. Under unweighted SINAD measurements, the stimulus signals were found to have a Signal In Noise And Distortion ratio of about 48dB, almost 2dB lower than the theoretical maximum. This difference is caused by non-quantization noise sources, such as thermal noise, and harmonic distortion. The SINAD measurements made for the systems tested under this project will have been degraded by the fact that the input signal itself had such low SINAD with respect to the manufacturer-quoted and theoretical maximum ratios for each system. Ideally, a high-purity sine wave generator would have been used for frequency sweep tests, causing minimal adverse impact on measurement values.

In addition to somewhat impure source signals, the ELVIS's data acquisition interfaces would also have had an impact on the accuracy of the measurements made. The analogue input channels sample data using 16-bit coding, and while the input voltage range was adjusted for maximum measurement resolution, the channels are susceptible to quantization noise. The culmination of these issues renders the measurements made somewhat unsuitable as absolutely accurate figures, and may account for discrepancies between my measurements and those presented by the manufacturers, but nonetheless they are still useful for characterisation and comparison of the various audio transmission technologies.

Another limitation of quantitative audio equipment metrics is that it can be argued that they do not yield meaningful indications as to the perceived quality of delivered sound, since the test methodologies use somewhat unrealistic input signals. Under everyday use, audio systems tend to have to deal with inputs much different than pure, individual sinusoidal tones. While the metrics calculated can be used to measure the ability of a system to handle the inputs applied under the tests, similar performance may not hold across the wide range of signals which will be encountered in 'real life'.

10.4 ITU-R 468 Noise Weighting Compliance and Suitability

As shown throughout Chapter 9 *Test Results & Analysis*, the application of the MATLAB-designed ITU-R 468 Noise Weighting filter to the stimulus and response signals captured during the Frequency Sweep tests has had a similar effect on the SINAD characteristic graphs for each system. An example of the effect of the filter in the time domain is given in Figure 10.3 below, which shows the power spectra for the Joyo JW-01 response to a 1kHz input with and without application of the noise weighting filter.

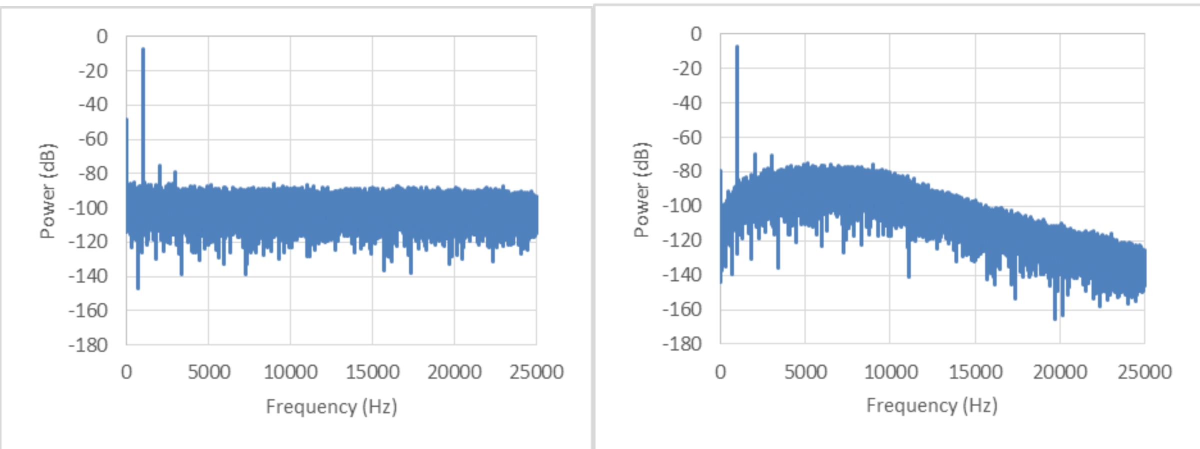

Figure 10.3 – Frequency-domain power spectra for the Joyo JW-01 response to a 1kHz input, both before (right-hand graph) and after (left-hand graph) ITU-R 468 Noise Weighting

The spectra of Figure 10.3 demonstrates the effect of the filter of boosting the frequencies around 6kHz and cutting others as described under Chapter 5 *Audio Quality Assessment*. The SINAD measurements in this case are 47.07dB before weighting and 24.16dB after weighting. While SINAD measurements throughout the analysis process of the project use traditional root-mean-square power averaging, it was discovered late in the project that the ITU-R 468 Recommendation requires quasi-peak measurement according to specific measurement characteristics defined by the Recommendation. Such measurement is not supported by LabVIEW, which defines SINAD as does equation 7.4, using root-mean-square measurement. As such the SINAD measurements conducted during the project are not directly comparable to external 468-weighted measurements, as they do not fully comply with the ITU Recommendation.

A further issue with the Noise Weighted SINAD measurements conducted during the project is that the weighting filter in each case altered the fundamental signal power in addition to the noise powers, thereby improving or degrading the SINAD ratio according to the frequency of the fundamental and the response of the filter. If SINAD measurements were to be made again, perhaps a better application of the Noise Weighting filter would be to calculate the ratio between the unweighted signal power to the noise and distortion power in the weighted signal. Measuring SINAD in this way would avoid the action of the filter on the fundamental signal power and yield more meaningful measurements by giving greater impact to the spectral composition of noise and distortion components.

11. Conclusions

11.1 Main Findings

This report has covered a wide range of topics related to the musical instrument cable replacement application, including wireless communications, audio quality assessment, measurement instrumentation, and frequency-domain analysis. It has been found that many methods exist for the assessment of audio quality, both quantitatively and qualitatively. Quantitative measurements such as Total Harmonic Distortion and Signal-to-Noise Ratio are often quoted by manufacturers of audio equipment as single figures, yet by calculating such metrics across the audible frequency range, this project has served to demonstrate that audio quality measurements are often variable across the range of operation, such that the use of a single measurement value in each case is unsuitable for characterising the performance of a system. As such, audio quality measurements are also highly dependent on the conditions under which the metrics are calculated, for example the fundamental frequency used for THD measurement.

Analogue technology, represented by the Samson Stage 5 VHF wireless system, while able to deliver very low latency, cannot do so without effecting the ‘attack’ of the signal, i.e. the speed at which a the signal reaches full amplitude from zero, due to the rise time characteristic of analogue filter circuitry. Other drawbacks of analogue communications in this domain include susceptibility to interference causing the loss of signal integrity, and limited frequency response which adversely affects the tone of the musical instrument. These problems can be somewhat improved upon by employing more expensive and advanced electronics, particularly in terms of robustness by the use of diversity antennae; a pair (or more) of antennae which allow for selection of the strongest and best signal, thereby improving the quality of output. Even without such enhancements, however, analogue systems tend to be already very expensive in comparison to their digital counterparts.

The digital systems investigated under this project have manifested a number of improvements over analogue technology, but also some serious limitations. Thanks to the mass-producibility of specialised chipsets these systems tend to be cheaper, and low power consumption allows for effective utilisation of rechargeable batteries. The digital systems proved to be much less susceptible to harmonic distortion than the analogue, but, being digital, their output suffers from the introduction of quantization noise. In terms of frequency response, the digital systems also have the advantage in that they delivered wider, flatter responses in the audible range.

Comparing the Laird DVK-BTM511 Bluetooth system to the Joyo JW-01's proprietary 2.4GHz implementation, a number of interesting differentiations were discovered. The lighter-weight proprietary 2.4GHz protocol proved more suitable to the instrument cable replacement application thanks to comparatively low latency, at the cost of decreased robustness and signal quality. The extra processing overhead required by Bluetooth yields greater robustness in an adverse environment, but also has the effect increasing the end-to-end latency introduced by the system to an unacceptable extent. The use of Bluetooth does however have the advantage of ensuring compatibility with a wide range of devices as eluded to in the Introduction.

Considering the impact of audio codecs over A2DP Bluetooth audio streaming link, the tests conducted indicate little advantage as to the use of the advanced aptX codec over standard SBC. While latency was slightly improved, it remained well outside the appropriate limit. The tests also showed no significant improvement in terms of frequency response, although manifested the impact of the DSP techniques associated with the aptX codec on the frequency composition of the signal, dependant on the composition of the input.

11.2 Suggestions for Further Research

In addition to those suggestions made regarding noise weighting and measurement instrumentation under Chapter 10 *Discussion*, there are a number of areas into which further research would be beneficial if the work completed under the project as documented by this report was to be continued. The computation of audio metrics not measured under this project, such as intermodulation distortion, may shed further light on the impact of different wireless communications systems on the quality of delivered audio.

Regarding Bluetooth, in addition to the SBC and aptX codecs investigated under the project, several other codecs are also available, such as AAC (Apple Advanced Codec) and aptX Low Latency, which may yield further measurable improvements on the suitability of Bluetooth for this application. It would also be useful to measure perceived audio quality delivered by the systems using realistic guitar audio samples by use of psychoacoustic modelling or MUSHRA test methodologies, as described under Chapter 5 *Audio Quality Assessment*.

11.5 Final Remarks

Through the completion of this project, a broad knowledge in the areas of wireless communications, audio quality assessment and the use of measurement instruments, particularly the ELVIS platform and its enhancement through the development of bespoke software tools using high-level programming languages, has been developed. Incorporating the LabVIEW and MATLAB languages a reliable and innovative test methodology was applied to experimentation and data analysis. Additionally, skills have been developed such as organisation, time management and the application of engineering knowledge.

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