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Mitigating Noise and Interference in Audio Signal during Virtual Meetings Using Audio Porting in Digital Signal Processing (APS)

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ABSTRACT *– Virtual Meetings has become part of our everyday activities and they are now part of our life because of the new-normal brought about by Covid-19 in the year 2020. Virtual meetings are held either on audio/video, video or audio modes and each of the modes are transmitted through a digital signal processing which produces noise and interference. This paper will demonstrate and apply signal processing methods through audio cable for noise reduction. This is necessary because virtual meetings can be a source of stress if the voice output coming from the audio systems is not clear and noiseless. It will identify various types of noise experienced during virtual meetings and propose a method of physical hardware integration to mitigate noise during a virtual presence. The study will also demonstrate ways in which local meeting hosts and participants can handle some hardware connections outside the methodology of this study and have a noiseless virtual centre. This paper will benefit users of various virtual applications, virtual application developers, schools, government organizations and others.*

Key Words*:* Digital Signal Processing, Audio Porting, Online Meeting, Audio Signal, Audio Distortion

I. Introduction

Organizations, large and small are increasingly embracing online meetings or virtual meetings held using an internet browser or computer application to link geographically dispersed workers, enhance productivity, and decrease travel costs. These applications can help you and your team hold more successful virtual meetings and boost cooperation if you have a better grasp of online meetings. (Cisco Webex 2020). Audio conferencing is frowned upon because of audio distortion. When meeting attendees are unable to hear the details of a conversation (and must ask for items to be repeated), frustration grows, and attention and productivity suffer. An audio conference that is disturbed by unwanted noises, echoes, or distortions is nearly impossible to beat. This is where the focus of this study is. The study examines the many sorts of noises encountered during online meetings, as well as what online meeting program developers have done to eliminate specific types of noise during online meetings, as well as additional ideas and implementations for noiseless online meetings.

Figure 1.0 presents the pattern in which audio signals are passed through a digital system.

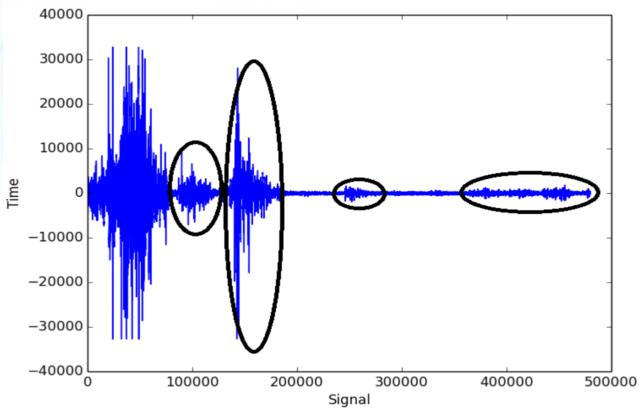

Figure 1.0: Audio Wobbling Sound with Increasing Voice Output.

Figure 1.0 shows an audio signal that has not been filtered either through an audio filtering application or hardware integration. As such the audio output keeps wobbling during the online meeting. The audio signal time for voice output is calculated at:

$$0+4000/0\text{-}4000$$
$$= 4000/\text{-}4000$$
$$= \text{-}1.$$

The above equation shows that when an audio output carries noise from the beginning of transmission to the destination, the result is equal to -1

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According to Cisco Webex (2020), online meetings are held using a web browser application or software installed on a PC or mobile device. Users can connect with one another via virtual meetings, also known as web conferences or video conferencing, using online meeting software. In most circumstances, all you need for an online meeting is a computer or mobile device with a microphone, online meeting software, an internet connection, someone to meet with who has the same components, and an agenda. Of course, if you choose to use sophisticated collaboration tools like web conferencing or video conferencing, both of which fall under the "online meeting" umbrella, you can enhance the functionality of your virtual conference. During meetings, web conferencing allows users to share their displays, poll attendees, whiteboard digitally, and more. Video conferencing incorporates a real-time video component to help develop team development and provide face-to-face communication in remote meetings.

Web conferencing, as described by Byrd, N. (2020), is an umbrella term for numerous types of online conferencing and collaborative services, such as webinars ("web seminars"), webcasts, and web meetings. In an attempt to distinguish it from the other forms of collaborative sessions, it is sometimes used in the more restrictive meaning of the peer-level online meeting setting. Apart from daily check-ins, planning sessions, and debriefings, online meetings can be a useful alternative to traditional in-person or audio-only gatherings. Nearly 77 percent of firms use virtual training to provide professional development for their staff, indicating that training is becoming more important. E-learning, webinars, and conferences are also popular.

1.1 Problem Statement

Vendors offering solutions for teams of various sizes and budgets abound in the online meeting sector. The capabilities you need from your online meeting software are universal, regardless of budget, for both large and small businesses. This study is focused on the mitigation of equipment noise and interference during a virtual meeting of a large gathering. The largest providers of conferences now have strong professional call quality. They use the latest technologies and route calls to the nearest data centre. However, there is a little use of modern accessible technology on certain free video conference sites. Older sites for online meetings could also use obsolete audio technologies. Another common issue with audio technology is that it is difficult to see who is on the dial. It is important to know who is with us and who talks and to know who participants are mute/unmute. Call quality is not a common issue for modern systems in video conferencing. This paper therefore applied some noise reduction techniques through use of cables and Ox ports on audio signal ports as a method of hardware integrations. It achieved this by evaluating the causes of different noise patterns during a virtual session, design a method of reduction through the use of Audio cable and ports (Audio Porting) and applied the design of noise reduction signal to enable online meeting attendees have a seamless online presence without nose interruption.

1.2 Various types of noise experienced during virtual conferences, meetings and calls includes:

- Noise caused by the wind
- Electrical Noise
- Interference Noise
- Ground Loops Noise pattern
- Rumble
- Mechanical Noises
- Equipment Self Noise
- Background Noise

2.0 Related Works

All meetings are built on the foundation of audio. Without good audio, the entire cooperation event can soon devolve into frustration and inefficiency. The digital signal processor (DSP) is critical in the conference room and has the ability to modify the audio experience, prompting significant study in digital signal processing of audio outputs during virtual meetings.

According to Lourent Ouder (2015), the most prevalent degradations in audio signals can be divided into two categories: localized and global. The global degradations affect all of the waveform's samples, whereas the localized degradations only affect particular groups of samples, resulting in a waveform discontinuity. Background noise, hiss, flutter, and certain types of non-linear degradations such as speed changes or distortion are examples of global degradations. Clicks, bursts, outliers, crackles, scratches, and other localized degradations are the subject of Lourent Ouder's (2015) research. The deterioration of ancient phonograph recordings is particularly widespread. Detection (identifying the locations of the degraded samples) and interpolation (restoring audio signals distorted by impulsive noise) are two difficult aspects in the process (replacing the degraded samples by more suitable values).

Tim Root (2016) presented a digital signal processing (DSP) method for reducing unwanted noise in virtual meetings, particularly noise caused by equipment and network interference. In a conferencing system, a DSP oversees the audio inputs and outputs, processing all audio signals to offer clear, intelligible speech to conference participants. A digital signal processing (DSP) can take numerous forms, according to Timi, the backbone of conference audio. This entails the integration of a variety of audio devices to deal with noise generated by interference or equipment noise. Tim Root is a writer who lives in the United (2016).

In managing noise through suppression method, Zoom (2020) developed a noise suppression feature and integrated such into the Zoom desktop client application for virtual meetings. Background noise, like paper crunching, keyboard typing, fan noise, dog barking, and other noises will be filtered out to create a better meeting experience. By default Zoom automatically does do background noise reduction, however the option can be changed to be more or less aggressive based on the environment and use case, Zoom Web (2020).

Liu et al. (2019) suggested a batteryless electrocardiogram (ECG) monitoring chip and intelligent gadget with wired audio transmission for long-term and real-time ECG monitoring. The ECG signal is modified and amplified in the front-end chip before being delivered over the 3.5-mm headphone cord's microphone channel. The front-end chip converts the sinusoidal signals from the left and right channels into a dc supply and a precise local oscillator signal, respectively. The suggested system samples the signal from smart devices using a high-precision audio analog-to-digital converter and processes it using an internal program. As a result, this gadget does not require an external battery, a local oscillator, or any other complicated modules. The usage of a chopper/amplitude modulation (AM) reused mixer amplifier, which combines chopping and AM while reusing the local oscillator signal, is suggested. The suggested amplifier's closed-loop gain ranges from 20 to 47 dB and may be automatically adjusted based on the output signal's amplitude. The suggested chip was made in a typical 0.18-m CMOS process and has a core area of 1.07 0.95 mm 2. The device is powered by a 1.5-V dc supply from the power management module. The chip consumes a total of 156 W when linked to the headphone wire load.

Good and quality audio is the answer to all multimedia services. Bernhard et al (2014), presented that the transmission of audio signal that relies on efficient algorithms of encoding and decoding (codecs) that reduce the channel capacity needed but, even when transmission errors happen, give still an excellent audio quality. Mp2, mp3, aac and ac3 are the most common audio codecs. The codecs use sophisticated algorithms that mimic hearing properties. Special things like high-frequency loss, narrow- band noise and pre-echoes can result from care. Ultimate consistency must be tested by statistically accurate hearing testing. ITU Guidelines P.800, BS.1116, and BS.1534 describe comprehensive protocols for performing accurate speech and audio tests. Instrumental methods for measurement like BS.1387 replicate subjective measures to approximate the perceived efficiency. The newest standardized approach is ITU Recommendation P.1201, which estimates the sound quality of the signal transmitted without a reference signal.

Young et al (2014) illustrated five barrier types in a real urban setting, and the acoustic characteristics of the five barrier types were expected. Laboratory studies have measured noise irritation, noise preconceptions about the barrier's output attenuation, the esthetic preference and the overall preference for noise barriers. There have been three distinct types of experiments: audio-only tests, only visual and audio-visual tests. The results of the experiments demonstrated a dominant effect

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on the perception of noise-attenuating efficiency of barriers by the reduction in noise, particularly with low frequency components. The noise barriers' bias premises influenced the overall noise barrier preference at 55dBA and the esthetic noise barrier preference at 65dBA. In addition, vegetation-based barriers have improved the perceived efficiency of noise barriers with increased esthetics and decreased noise perceptions- Young et al (2014).

Wei et al (2017) have proposed a new approach based on empirical mode set (EEMD) and threshold filtering of wavelets for random noise. Proposed First of all, Fourier's spectrum analysis approach examines the frequency band ranges of effective signal and noise in the initial seismic data collection. Secondly, to obtain the IMFs of the original noisy trace, we use the EEMD process. The wavelet thresholds are then used to acquire new denoted high-frequency IMFs for the IMFs of each track. Finally, we can obtain the ultimately denoted seismic data set by stacking the filtered high frequency IMFs with the low frequency IMFs and the trend item. Two synthetic data examples and one field data example are used to validate the method proposed. The results show that the method proposed can produce much cleaner denouncing efficiency without bias to most useful signals. Wei et al (2017).

A Review of Existing Audio/Video Conference Application
-Zoom Online Conference Application:

Noise suppression is a feature in the Zoom client that can assist reduce distracting noises picked up by participants' microphones. To improve the meeting experience, background noise such as paper crunching, keyboard tapping, fan noise, dog barking, and other noises will be filtered away. Zoom automatically reduces background noise by default; however, depending on the setting and use case, the setting can be altered to be more or less aggressive.

Installation requirements for the Zoom Desktop Client for Windows program Zoom Desktop Client for Windows, version 5.2.0 or higher

Version 5.2.0 or higher of the Zoom Desktop Client for macOS

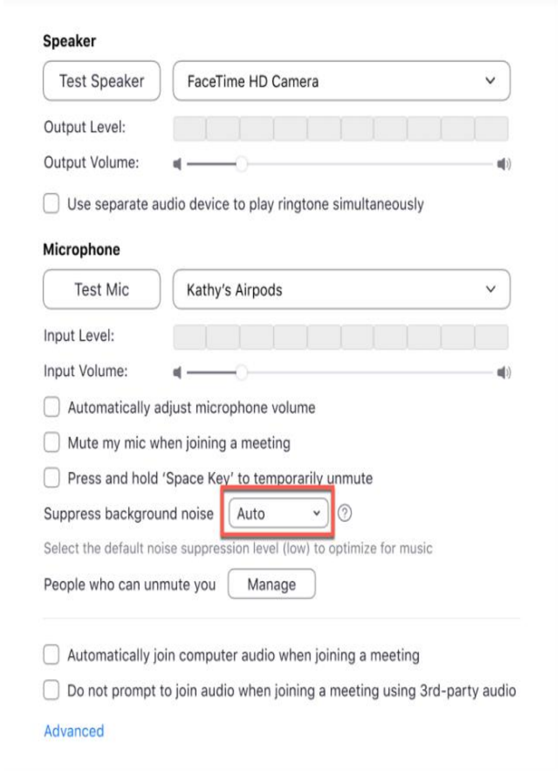

Figure 2.1: Architecture of Zoom Noise reduction method (Zoom, 2021)

Detailed instructions

Click your profile image in the Zoom Desktop Client, then Settings.

Click the Audio tab.

Click the drop-down menu next to Suppress background noise, to change the setting:

Auto: This is the default setting, you can, when necessary, use a moderate amount of background noise reduction. Based on what it detects in the background, it will automatically alter the aggressiveness for blocking background noise. It will not treat music as background noise if it is detected.

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The Zoom Noise suppression feature has the following setups:

Low: There will be very little noise reduction. Low amounts of persistent background noise will be blocked.

Note: This setting is best for listening to music casually because it preserves as much of the original sound as possible. Consider using the Enable Original Audio setting in your advanced audio settings to get the best sound quality when playing music.

Medium: Ideal for reducing and eliminating background noise in typical settings, such as fans, pen tapping, and so on.

High: Noise reduction will be at its most aggressive, and noises like crunching of paper or wrappers, keyboard typing, and so on will be eliminated.

Note: Enabling this option may result in increased CPU usage.

Microsoft Teams Online Conference Application:

Microsoft also integrated noise cancelation as a feature in Microsoft Teams. Microsoft is the developer of Microsoft Teams. Microsoft Teams is one among the generally acceptable audio/video online meeting application.

Set the noise suppression level

You can change this setting at any time. Once changed, the setting carries over to your next meeting or call.

To affect the noise suppression level for a meeting you're currently in, use the second procedure.

From the main Teams window

1. Select your profile picture at the top right of Teams and then select Settings.
2. Select Devices on the left and then, under Noise suppression, select an option.

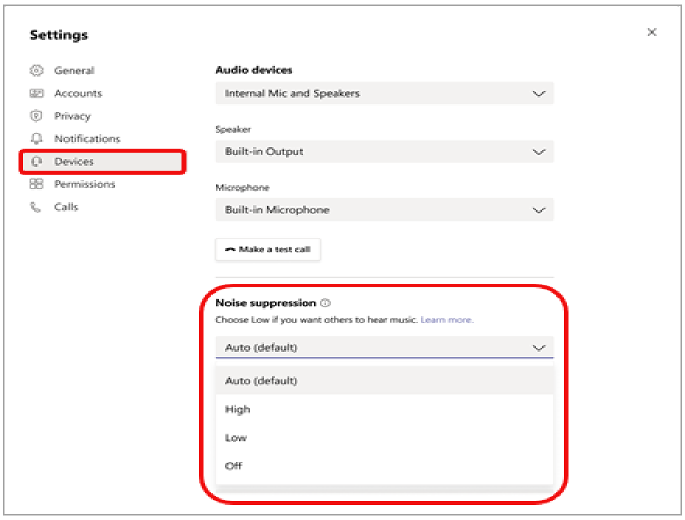

Figure 2.2: Team Architecture for Noise Suppression (Microsoft, 2021)

Teams' app determines the appropriate amount of noise reduction. The following Settings are contained in Microsoft Team Meeting Settings.

• **High**: Suppresses all non-speech background noise.

Observations:

Advanced Vector Extensions 2 must be supported by your computer's processor for this option to work (AVX2).

If the meeting or call is being recorded or if live captions are enabled, this option is presently unavailable.

Allowing this option to be enabled consumes additional computer resources.

• **Low:** Suppresses low levels of persistent background noise, such as a computer fan or air conditioner. Use this setting for playing music.

• **Off:** Noise suppression is disabled. Use this setting for high-fidelity microphones in low noise environments

From a meeting window

1. Select more options ••• in your meeting controls and then select Device settings.
2. Under Noise suppression, select an option. (See step Device Settings windows for more settings.)

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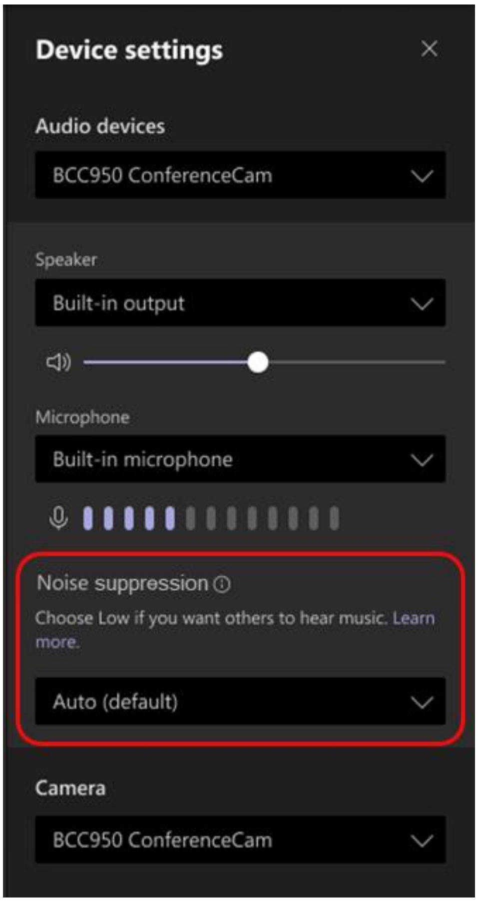

Fig. 2.3

3.0 Methodology

This work will adopted a rapid application development method. Because, with rapid application development, the user is involved in all phases of the life cycle, not only requirements definition, but design, development, test and final developments as well. Although no system application is developed for this work, an audio signal processing was done using audio cables to ascertain the way to mitigate noise during virtual sessions. Rapid applications ensured that there were quick turnaround during the connection and text sessions. User involvement also increased during the processes.

4.0 Method

4.1 Audio Porting

The sound hardware of a computer is connected to your speaker, the microphone, headsets and other audio equipment through audio connections. At least few simple audio ports are built in for each computer motherboard, which allow connection of a stereo and a microphone. Conference devices typically have integrated digital signal processor (DSP) chips that provide the processing power and other technologies required to create a high-quality experience. Larger collaboration environments, which use a bigger number of standalone microphones and loudspeakers and also require a dedicated DSP appliance such as Ox Ports linkers, Intellimix, and so on, are given more emphasis in this study.

The processing capacity required for the multiple audio inputs and outputs in the room and throughout the unified communication (UC) network is provided by this discrete unit. The first job of Audio Porting (AP) when meeting participants connect via audio endpoints is usually to execute acoustic echo cancellation (AEC). Someone else's communication equipment has failed to echo-cancel your signal if you hear your own voice echo in the nearby speaker. Your voice is picked up by the far-end microphone(s) and played through your speaker as it is amplified on the far end (the meeting area you're linked to). Before transmitting the signal on, AEC hardware and software compare the microphone's signal to your own incoming voice being played out on the other end, and remove it from the microphone's signal. An AP provides automatic gain control once echo has been removed (AGC). Each receiving microphone signal should be amplified to about the same level by the AP. Users on the other end of the line are either blasted by a loud individual or strain to hear a softly spoken individual without AGC. Throughout the conversation, AGC adjusts the loudness of voices to a comfortable level.

Mixing and gating technology are also provided by audio porting in a digital signal processing method. If all of the microphones must feed into a single output (such as a phone line or a videoconferencing device), the input signal from each microphone must be mixed into the output signal. The outgoing signal(s) may be routed through several outputs, such as the internet, a phone line, or a recording system, simultaneously or independently. The term 'gating' refers to the system's ability to

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deactivate particular microphones selectively. Some powerful AP algorithms, for example, can tell the difference between human speech and noises like typing, paper crinkling, and finger tapping, and disregard signals from a loud mic until speech is identified again.

ASP definition and encoding are required for output signals. As a result, the audio porting signal outputs analogue signals to speakers or an amplifier, as well as a digital stream to a VoIP phone system or a digital stream to USB or Ethernet ports (AVB or Dante formats, for example). The ASP must offer the right digital signal coding/decoding (using a codec) of industry-standard particular formats required by the channel and protocol because the microphone signals may be mixed or unmixed. The ASP and codec transform analogue speech signals to digital and back in conference audio speech for digital systems. The amount of data that can be transmitted digitally increases the amount of traffic that can be carried through a network. By encrypting the data stream, the ASP and codec can provide security while also potentially decreasing required transmission bandwidth.

Finally, an ASP offers additional features to enhance the user experience. The noise reduction capabilities in the microphone pickup signal minimizes ambient noise, preventing it from being heard on the other end. Noise fill, on the other hand, injects noise so that a listener does not believe he is being disconnected when the environment is too quiet and no one is speaking. For larger, more sophisticated systems, master mute and volume control provide even more intelligence.

Whether the conferencing application is small or large, the audio signals going in and out of the room are fine-tuned and optimized by an APS. As a result, everyone can hear clear, coherent communication, allowing for successful, fruitful teamwork.

4.2 Illustrated Audio Output Transmitted with Echo and Background Noise

The Following figures demonstrate the point at which audio signal was lost and the percentage of the final transmission. This demonstrates the improvement of Audio Porting for improved audio output during virtual presence.

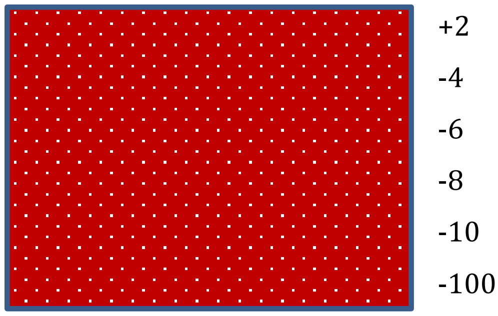

Figure 4.2.1 Audio transmission with 40% echo and feedback

Figure 4.2.1 shows that audio signal was received, But not without noise. The noise input is frequent and visible that the expected sound. The audio output was lost at -4 which is targeted at 55%.

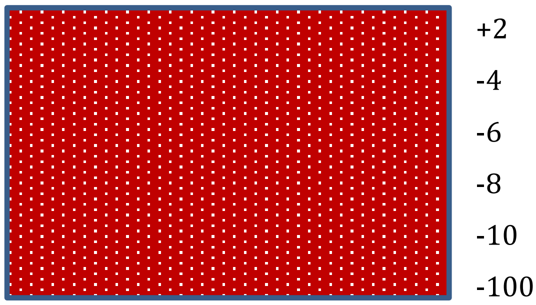

Figure 4.2.2 Audio transmission with 55% echo and feedback.

Figure 4.2.2 shows that audio signal was received, but not without noise. The noise input is frequent and visible that the expected sound. The audio output was lost at -6 which is targeted at 55%.

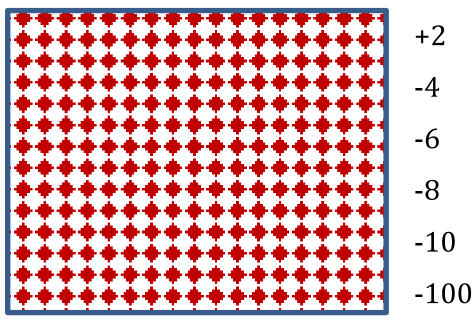

Figure 4.2.3 Audio transmission with 80% echo and feedback

The above illustrations shows that audio signal was received, but not without noise. The noise input is frequent and visible that the expected sound. The audio output was lost at -8 which is targeted at 80%.

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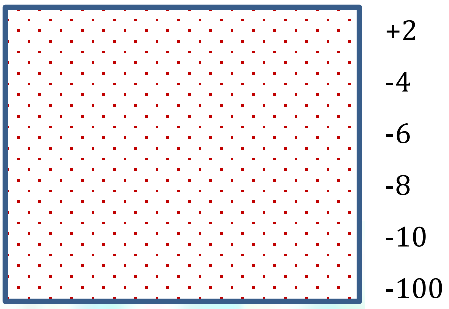

Figure 4.2.4 Audio transmission with 10% noise

Figure 4.2.4 shows that audio signal was received with an improved audio with 10% noise. The noise input is not frequent and there was clarity of voice output from the audio system.

5.0 Results and Discussion

The Audio Porting Digital Signal Processing is a combination of audio mixing and gating devices, such as OX cables and IntelliMix DSP (DSP). This unit's patented algorithms increase every aspect of conference sound, providing you and your team with the highest possible audio quality for video conferencing in any size room.

Meeting participants and hosts are spared the agony of bad audio caused by extraneous noise or distortions thanks to APS. Rather, it provides unrivaled clarity to improve communication, workflow, and, as a result, corporate efficiency. The only interruptions come from a natural flow of brilliant ideas, not the obstacles associated with bad audio, when meeting participants hear every detail.

5.1 Usability of Audio Porting Digital Signal Processing

The P300 Audio Conferencing Processor and the Microflex Advance MXA910 Ceiling Array Microphone are equipped with ASP portfolio of digital signal processing algorithms. Figure 5.1 shows the usability and flexibility of Audio Porting Digital Signal Processing.

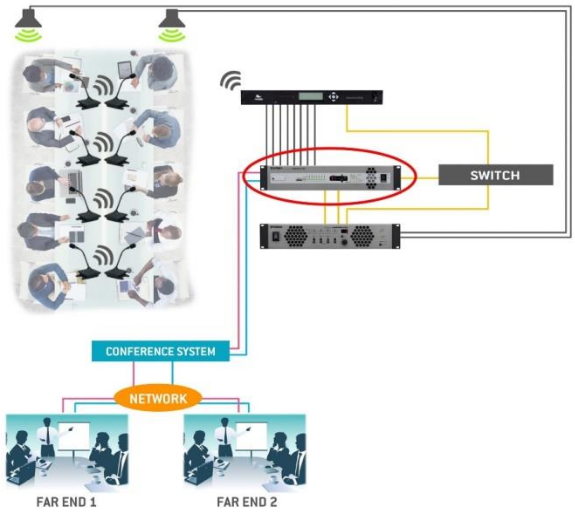

Figure 5.1: Audio Porting Digital Signal Processing

As shown in figure 5.1, the Microflex Advance MXA910 Ceiling Array Microphone with IntelliMix combines state-of-the-art array microphone technology with the Shure IntelliMix digital signal processing. It offers - Acoustic Echo Cancellation, Noise Reduction, and Automatic Mixing for exceptional and unprecedented control.

The microphone features outstanding "ease of deployment" capabilities. These include simplified templates and Autofocus Technology for immediate out-of-box setup. Designer software compatibility provides unmatched ease of configuration, so the system can be installed quickly and efficiently.

5.2 Features and Benefits of Audio Porting Digital Signal Processing

1. Acoustic Echo Cancellation (AEC) eliminates far-end audio leakage into the microphone, resulting in an echo-free conference.
2. Noise Reduction digital signal processing aids in overcoming difficult space acoustics by reducing unwanted distractions like HVAC noises, which can obstruct clear conversation.
3. Automatic mixing improves clarity, intelligibility, and provides seamless interaction between conference venues by increasing presence and reducing noise pickup and transmission.

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4. Automatic Gain Control (AGC) is a sort of digital signal processing that adjusts the volume on each channel to ensure that quiet and loud talkers are heard at the same time.
5. Compression smoothes out abrupt fluctuations in the automixer's output signal for better audio quality.
6. When a video system introduces latency, delay aligns audio and video (where you hear someone speak, and their mouth moves later).
7. PEQ (Parametric Equalization) allows for accurate sound management. Every channel of each array microphone has a 4-band PEQ. Three EQ Contour settings are included with the MXA910 for quick and easy microphone tuning.
8. Enhancing the Audio Experience with Intelligence
9. IntelliMix digital signal processing combines several MXA910s with just one P300 processor to give better audio quality than ever before in larger meeting venues.
10. Combining these two items creates the optimum audio experience for conferences. Thanks to IntelliMix and the distributed DSP architecture, this scalability is easier to implement and less expensive. For multi-room installations or rooms requiring more than one ceiling array microphone, distributing DSP among microphones and communication interfaces reduces cost and complexity.

6.0 Conclusion

Noise causes stress and kills the interest of virtual applications users on the use of such applications for virtual sessions. To create a noiseless session when online, the recommendations in this study is a support to having seamless and noiseless virtual session. Although this work does not cover all patterns of noise control, it is efficient for background noise, noise caused by the wind, and noise as a result of interference. The study has also demonstrated ways in which local meeting hosts and participants can handle some hardware connections during virtual sessions despite the virtual application being used for the online presence. If the suggestions in this study are applied properly, it will improve and enhanced the online sessions held every day especially in the use of audio systems.

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