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FORENSIC TESTING PROGRAM

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# Forensic Audio Analysis
# Test No. 23-5591 Summary Report

Participants were provided with an audio evidence file. They were asked to examine the evidence utilizing their own tools and methods. Results were returned from 20 participants and are compiled in the following tables:

| | Page |
|---|---|
| Manufacturer's Information | 2 |
| Summary Comments | 4 |
| Table 1: Forensic Audio Examination Responses | 5 |
| Table 2: Forensic Audio Enhancement Responses | 12 |
| Table 3: Forensic Audio Enhancement Observations | 21 |
| Table 4: Additional Comments | 23 |
| Appendix: Data Sheet | |

This report contains the data received from the participants in this test. Since these participants are located in many countries around the world, and it is their option how the samples are to be used (e.g., training exercise, known or blind proficiency testing, research and development of new techniques, etc.), the results compiled in the Summary Report are not intended to be an overview of the quality of work performed in the profession and cannot be interpreted as such. The Summary Comments are included for the benefit of participants to assist with maintaining or enhancing the quality of their results. These comments are not intended to reflect the general state of the art within the profession.

Participant results are reported using a randomly assigned "WebCode". This code maintains participant's anonymity, provides linking of the various report sections, and will change with every report.

# Manufacturer's Information

The Forensic Audio Analysis test consisted of an evidence audio file. Participants were asked to enhance the audio file to minimize distracting elements, and clarify the speech from the incident contained in the file.

SAMPLE PREPARATION: A predetermined, staged event involving a traffic stop with an individual refusing commands from multiple officers is captured on a body camera system with a dying battery. Audible beeping from the dying battery covers parts of the speech. The .wav file was zipped and uploaded to the CTS Portal for participants to download. A MD5 and SHA1 hash value was calculated and provided for the compressed file to allow participants to validate the successful download of the file.

SAMPLE VALIDATION/VERIFICATION: The combination of internal test validation and the responses received from the predistribution laboratories confirmed the expected responses. The following list of tools were utilized in the validation of this test: iZotope RX advanced, ProTools, Audacity, Quickhash, Mediainfo. CTS does not endorse any particular tools.

SCENARIO PROVIDED TO PARTICIPANTS

An audio file has been submitted to you for examination. You are asked to minimize the distracting elements, and clarify the speech from the incident contained in the file, so that the subjects can be better heard and understood.

# Manufacturer's Information, continued

## Question Manufacturer's Expected Response - Examination Questions

1-1 What is the SHA1 hash value of the audio file?

***Expected Response:***

5b22c920efc7d0aa9a3b3d147354d21b6f66278a

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1-2 What is the file format of the audio file?

***Expected Response:***

WAVE / .wav / PCM

---

1-3 What is the sample rate of the audio file?

***Expected Response:***

48 kHz

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1-4 What is the bit depth of the audio file?

***Expected Response:***

24 Bit

---

1-5 Is the audio signal clipping?

***Expected Response:***

Yes

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1-6 Starting with the lowest frequency (i.e. Frequency 1) and moving upward, what are the four main frequencies (or frequency ranges) that make up the beeps / audio prompts that repeat every 10 seconds?

***Expected Response:***

Frequency 1: 200 Hz or 100-300 Hz
Frequency 2: 600 Hz or 500-700 Hz
Frequency 3: 4 kHz or 4,000 Hz or 3900-4100 Hz
Frequency 4: 8 kHz or 8,000 Hz or 7900-8100 Hz

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## Question Manufacturer's Expected Response - Enhanced Audio Examination

2-1 Note the methods or tools used and the settings for the audio enhancement here.

***Expected Response:***

This was a free form question on methods and tools used. No manufacturer's response expected.

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# Summary Comments

This test was designed to allow participants to assess their proficiency in data verification, media characterization, data analysis, signal analysis and enhancement of an audio file using their own tools and methods. The participants were provided with an audio file and were asked to answer questions as well as make enhancements to the audio file. (See Manufacturer's Information for preparation details, test scenario, and test questions)

A total of 20 participants returned results for this test. A variety of software tools were used by participants during their examination. The most frequently reported tools included Adobe Audition and Izotope RX.

All examinations questions achieved consensus responses greater than 95%. In a separate section of this test, participants were ask to perform specific enhancement steps to the audio file and submit these enhanced audio files to CTS. An expert reviewed the audio files submitted by participants and provided observational notes in Table 3.

# Forensic Audio Examination Responses

TABLE 1

**Question 1- 1 : Examination Questions**

Question 1-1: What is the SHA1 hash value of the audio file?

Manufacturer's Expected Response: 5b22c920efc7d0aa9a3b3d147354d21b6f66278a

| WebCode | Response |
|---|---|
| 2PMJJE | 363a79eab2c148a70f8c4ba06a54ffdcf3ea6e4e97c9e0000dc366af98f22248 |
| 3XW9WD | 5B22C920EFC7D0AA9A3B3D147354D21B6F66278A |
| 62ARUD | 5B22C920EFC7D0AA9A3B3D147354D21B6F66278A |
| 69VK7C | 5B22C920EFC7D0AA9A3B3D147354D21B6F66278A |
| 6D3DZ9 | 5b22c920efc7d0aa9a3b3d147354d21b6f66278a |
| 6WHM29 | 5B22C920EFC7D0AA9A3B3D147354D21B6F66278A |
| 8YFKP7 | 5B22C920EFC7D0AA9A3B3D147354D21B6F66278A |
| BBGFA7 | 5b22c920efc7d0aa9a3b3d147354d21b6f66278a |
| DQ4H6Z | 5b22c920efc7d0aa9a3b3d147354d21b6f66278a |
| EPR3BY | 5B22C920EFC7D0AA9A3B3D147354D21B6F66278A |
| FUF23Z | 5b22c920efc7d0aa9a3b3d147354d21b6f66278a |
| HLFNVV | 5b22c920efc7d0aa9a3b3d147354d21b6f66278a |
| MHZKBU | 5b22c920efc7d0aa9a3b3d147354d21b6f66278a |
| MWGHFV | 5B22C920EFC7D0AA9A3B3D147354D21B6F66278A |
| Q2JNJQ | 5b22c920efc7d0aa9a3b3d147354d21b6f66278a |
| RVNFDN | 5b22c920efc7d0aa9a3b3d147354d21b6f66278a |
| V2M4JK | 5B22C920EFC7D0AA9A3B3D147354D21B6F66278A |
| WAWTWJ | 5b22c920efc7d0aa9a3b3d147354d21b6f66278a. |
| XMZB9J | 5b22c920efc7d0aa9a3b3d147354d21b6f66278a |
| YZKD4F | 5b22c920efc7d0aa9a3b3d147354d21b6f66278a |

Question 1-1: What is the SHA1 hash value of the audio file?

Consensus Result: 5b22c920efc7d0aa9a3b3d147354d21b6f66278a

TABLE 1

**Question 1- 2 : Examination Questions**

Question 1-2: What is the file format of the audio file?

Manufacturer's Expected Response: WAVE / .wav / PCM

| WebCode | Response |
|---|---|
| 2PMJJE | WAVE |
| 3XW9WD | wav |
| 62ARUD | wave /PCM |
| 69VK7C | PCM wav |
| 6D3DZ9 | Waveform Audio (wav) |
| 6WHM29 | WAC PCM |
| 8YFKP7 | WAV |
| BBGFA7 | RIFF/WAV |
| DQ4H6Z | wave |
| EPR3BY | WAV |
| FUF23Z | WAV |
| HLFNVV | wav |
| MHZKBU | WAV |
| MWGHFV | WAV |
| Q2JNJQ | Wav PCM |
| RVNFDN | WAV |
| V2M4JK | WAV(PCM) |
| WAWTWJ | PCM WAV |
| XMZB9J | Wave PCM |
| YZKD4F | WAV PCM 2-channel |

Question 1-2: What is the file format of the audio file?

Consensus Result: WAV (wave) and/or PCM

TABLE 1

**Question 1- 3 : Examination Questions**

Question 1-3: What is the sample rate of the audio file?

Manufacturer's Expected Response: 48 kHz

| WebCode | Response |
|---|---|
| 2PMJJE | 48.0 |
| 3XW9WD | 48.0KHz (48,000Hz) |
| 62ARUD | 48 kHz |
| 69VK7C | 48 kHz |
| 6D3DZ9 | 48.0 kHz |
| 6WHM29 | 48 kHz |
| 8YFKP7 | 48000 Hz |
| BBGFA7 | 48,000 samples per second |
| DQ4H6Z | 48 kHz |
| EPR3BY | 48 kHz |
| FUF23Z | 48 kHz |
| HLFNVV | 48 kHz |
| MHZKBU | 48000 |
| MWGHFV | 48,000 Hz |
| Q2JNJQ | 48000 Hz |
| RVNFDN | 48 kHz |
| V2M4JK | 48000 Hz |
| WAWTWJ | 48 kHz |
| XMZB9J | 48 kHz |
| YZKD4F | 48 kHz |

Question 1-3: What is the sample rate of the audio file?

Consensus Result: 48 kHz or 48000 Hz

TABLE 1

**Question 1- 4 : Examination Questions**

Question 1-4: What is the bit depth of the audio file?

Manufacturer's Expected Response: 24 Bit

| WebCode | Response |
|---|---|
| 2PMJJE | 24 |
| 3XW9WD | 24bits |
| 62ARUD | 24 - bit |
| 69VK7C | 24 bits |
| 6D3DZ9 | 24 bit |
| 6WHM29 | 24 bit |
| 8YFKP7 | 24 |
| BBGFA7 | 24-bit integer |
| DQ4H6Z | 24 bits |
| EPR3BY | 24 bits |
| FUF23Z | 24 bits |
| HLFNVV | 24 bit |
| MHZKBU | 24 |
| MWGHFV | 24 |
| Q2JNJQ | 24 bits |
| RVNFDN | 24-bit |
| V2M4JK | 24 bit |
| WAWTWJ | 24-bit |
| XMZB9J | 24 |
| YZKD4F | 24 bit |

Question 1-4: What is the bit depth of the audio file?

Consensus Result: 24 Bit

TABLE 1

## Question 1- 5 : Examination Questions

Question 1-5: Is the audio signal clipping?

Manufacturer's Expected Response: Yes

| WebCode | Response |
|---|---|
| 2PMJJE | Yes |
| 3XW9WD | yes |
| 62ARUD | Yes |
| 69VK7C | Yes - the waveform is squared off on several occasions at 0dBFS resulting with audible clipping |
| 6D3DZ9 | Yes |
| 6WHM29 | Yes |
| 8YFKP7 | yes |
| BBGFA7 | YES |
| DQ4H6Z | yes |
| EPR3BY | Yes |
| FUF23Z | Yes |
| HLFNVV | yes |
| MHZKBU | YES |
| MWGHFV | Yes |
| Q2JNJQ | Yes, the audio signal is clipping. |
| RVNFDN | Yes |
| V2M4JK | Yes, audible and visual clipping can be found at the following time intervalls; 38 s 816 ms - 40s 000 ms; 41 s 496 ms - 43 s 402 ms; 56 s 000 ms - 56s 727 ms; 01 min 01 s 401 ms - 01 min 03 s 589 ms; 01 min 07 s 119 ms - 01 min 07 s 506 ms; 01 min 16 s 937 ms - 01 min 17 s 249 ms |
| WAWTWJ | Yes signal clipping detected at clicks and can be heard at speech at 40sec and 1min marks. |
| XMZB9J | yes |
| YZKD4F | Yes |

Question 1-5: Is the audio signal clipping?

Consensus Result: Yes

TABLE 1

## Question 1- 6 : Examination Questions

Question 1-6: Starting with the lowest frequency (i.e. Frequency 1) and moving upward, what are the four main frequencies (or frequency ranges) that make up the beeps / audio prompts that repeat every 10 seconds?

Manufacturer's Expected Response: Frequency 1: 200 Hz or 100-300 Hz
Frequency 2: 600 Hz or 500-700 Hz
Frequency 3: 4 kHz or 4,000 Hz or 3900-4100 Hz
Frequency 4: 8 kHz or 8,000 Hz or 7900-8100 Hz

| WebCode | Response |
|---|---|
| 2PMJJE | Frequency 1: 202.28Hz,<br>Frequency 2: 599.09 Hz,<br>Frequency 3: 4018.71 Hz,<br>Frequency 4: 7976.42 Hz |
| 3XW9WD | Frequency 1: 170-230Hz,<br>Frequency 2: 560-640Hz,<br>Frequency 3: 3950-4050Hz,<br>Frequency 4: 7950-8050Hz |
| 62ARUD | Frequency 1: ~200 Hz,<br>Frequency 2: ~600 Hz,<br>Frequency 3: ~4000 Hz,<br>Frequency 4: ~8000 Hz |
| 69VK7C | Frequency 1: Centre frequency 200Hz,<br>Frequency 2: Centre frequency 600Hz,<br>Frequency 3: Centre frequency 4kHz,<br>Frequency 4: Centre frequency 8kHz |
| 6D3DZ9 | Frequency 1: 200 Hz,<br>Frequency 2: 600 Hz,<br>Frequency 3: 4000 Hz,<br>Frequency 4: 8000 Hz |
| 6WHM29 | Frequency 1: 200Hz +/- 2Hz,<br>Frequency 2: 600Hz +/- 2Hz,<br>Frequency 3: 4kHz +/- 10Hz,<br>Frequency 4: 8kHz +/- 10Hz |
| 8YFKP7 | Frequency 1: approx. 200 Hz,<br>Frequency 2: approx. 600 Hz,<br>Frequency 3: approx. 4 kHz,<br>Frequency 4: approx. 8 kHz |
| BBGFA7 | Frequency 1: 200 Hz,<br>Frequency 2: 600 Hz,<br>Frequency 3: 4,000 Hz,<br>Frequency 4: 8,000 Hz |
| DQ4H6Z | Frequency 1: 200 Hz,<br>Frequency 2: 600 Hz,<br>Frequency 3: 4000 Hz,<br>Frequency 4: 8000 Hz |
| EPR3BY | Frequency 1: 200 Hz,<br>Frequency 2: 600 Hz,<br>Frequency 3: 4000 Hz,<br>Frequency 4: 8000 Hz |
| FUF23Z | Frequency 1: 200 Hz,<br>Frequency 2: 600 Hz,<br>Frequency 3: 4000 Hz,<br>Frequency 4: 8000 Hz |

TABLE 1

| Question 1- 6 : Examination Questions | |
|---|---|
| **WebCode** | **Response** |
| HLFNVV | Frequency 1: Around 188 Hz,<br>Frequency 2: 656 Hz,<br>Frequency 3: 4031 Hz,<br>Frequency 4: 8063 Hz |
| MHZKBU | Frequency 1: 100-330,<br>Frequency 2: 520-750,<br>Frequency 3: 3880-4170,<br>Frequency 4: 7880-8200 |
| MWGHFV | Frequency 1: 174.70 - 226.89 Hz,<br>Frequency 2: 597.15 - 625.53 Hz,<br>Frequency 3: 3950.5 - 4049.7 Hz,<br>Frequency 4: 7970.2 - 8036.9 Hz |
| Q2JNJQ | Frequency 1: 200 Hz,<br>Frequency 2: 600 Hz,<br>Frequency 3: 4000 Hz,<br>Frequency 4: 8000 Hz |
| RVNFDN | Frequency 1: 200 Hz,<br>Frequency 2: 600 Hz,<br>Frequency 3: 4000 Hz,<br>Frequency 4: 8000 Hz |
| V2M4JK | Frequency 1: approximately 200 Hz +/- 2 Hz,<br>Frequency 2: approximately 600 Hz +/- 2 Hz,<br>Frequency 3: approximately 4000 Hz +/- 1 Hz,<br>Frequency 4: approximately 8000 Hz +/- 1 Hz |
| WAWTWJ | Frequency 1: 200Hz,<br>Frequency 2: 600Hz,<br>Frequency 3: 4kHz,<br>Frequency 4: 8kHz |
| XMZB9J | Frequency 1: 0-264,<br>Frequency 2: 514-692,<br>Frequency 3: 3705-4319,<br>Frequency 4: 7468-8430 |
| YZKD4F | Frequency 1: 200Hz,<br>Frequency 2: 600Hz,<br>Frequency 3: 4kHz,<br>Frequency 4: 8kHz |

Question 1-6: Starting with the lowest frequency (i.e. Frequency 1) and moving upward, what are the four main frequencies (or frequency ranges) that make up the beeps / audio prompts that repeat every 10 seconds?

Consensus Result: Frequency 1: 200 Hz or a range of values that includes this frequency
Frequency 2: 600 Hz or a range of values that includes this frequency
Frequency 3: 4000 Hz or a range of values that includes this frequency
Frequency 4: 8000 Hz or a range of values that includes this frequency

# Forensic Audio Enhancement Responses

TABLE 2

## Question 2- 1 : Enhanced Audio Examination

Question 2-1: Note the methods or tools used and the settings for the audio enhancement here.

Manufacturer's Expected Response: This was a free form question on methods and tools used. No manufacturer's response expected.

| WebCode | Response |
|---|---|
| 2PMJJE | Cedar Cambridge Audio suite used to carry out test. Direct file input. Retouch volume tool used to remove clicks and impulses in the first 20 seconds, then adjusted to remove the beeps. FNR (Forensic Noise Reduction) tool used to reduce the ambient noise, and the Adaptive Limiter tool used to add gain and limit the ceiling level to -1dBFS. |
| 3XW9WD | I used the Audacity program (ver 3.3.3) to analyze the provided file, 23-5591_Audio.wav. First, I identified and removed four instances of digital click sounds within the 0 to 20-second range. To reduce the recurring beep signal occurring every 10 seconds, I examined the relevant section and its frequency range. I then used the "Filter Curve Equalizer" in the Effects menu to decrease the -54dB level within the frequency range of the beep sound. Next, to improve the noise ratio between the speaker's voice and the background noise, I employed the "Noise Reduction" feature in the Effects menu. Since the speaker's voice is quiet in the initial part of the conversation, from 0 seconds to 33 seconds, it's necessary to increase the sound volume. I used the "Amplify" function in the Effects menu to boost the sound by 12dB and allowed clipping. Additionally, I increased the range of 230-560Hz and 640-3950Hz by 12dB using the "Filter Curve Equalizer" function. Then, I used the “Noise Reduction” function again to remove noise. I applied the "Normalize" function in the Effects menu to set the peak amplitude to –1.0dB. To export the edited audio, I selected the WAV extension and chose "Signed 24-bit PCM" for encoding. |
| 62ARUD | Sound Forge 10 (dc offset, hard limiter, compressor limiter, normalize). Izotope RX6 (EQ & Gain modules - using the frequency selection tool & minus gain as a notch filter) |

TABLE 2

| Question 2- 1 : Enhanced Audio Examination | |
|---|---|
| **WebCode** | **Response** |
| 69VK7C | The wav file was processed in iZotope RX 10 and Cedar Cambridge v.12.<br>iZotope RX 10:<br>- 'Mixing' module to select left-channel only.<br>- 'De-clip' module to reduce clipping distortion. Threshold (-0.3), Quality (High), Makeup gain [dB] (-3.0), Post-limiter (selected).<br>- 'De-click' module to reduce the intensity of the four digital clicks in the first 20 seconds of the audio file. Algorithm (Multi-band (random clicks)), Sensitivity (8.0), Frequency skew (0.0), Click widening [ms] (0.0). Rendered timings (00:00:05.195 – 00:00:05.224, 00:00:06.742 – 00:00:06.767, 00:00:13:936 – 00:00:13.963, 00:00:19.978 – 00:00:20.023).<br>- 'Spectral Repair' module 'Attenuate' function to reduce the intensity of beeping sounds that occur every ten seconds. Bands (512), Multi-resolution (selected), Surrounding region length [%] (100), Strength (2.0), Before/after weighting (0.0), Direction of interpolation (Vertical). Used spectrogram view and time-frequency selection tool in order to render only the four frequency ranges that make up the beeping sounds, during only the time periods they are present.<br>- 'De-click' module to remove residual clicks heard at beginning and end of sections where beeping sounds occurred. Algorithm (Multi-band (random clicks)), Sensitivity (2.0), Frequency skew (0.0), Click widening [ms] (0.0).<br>- 'De-hum' module to reduce tonal background noise. Adaptive mode (selected), Filter type (Dynamic), Sensitivity (2.7), Bands (130), Filter Q (720).<br>- 'De-click' module to further reduce intensity of clicks heard between 00:43-00:45 (mm:ss). Algorithm (Multi-band (random clicks)), Sensitivity (8.0), Frequency skew (3.8), Click widening [ms] (1.2). Rendered timings (00:00:44.326 – 00:00:45.029).<br>- 'De-wind' module to reduce wind noise between 00:00:54.000 – 00:01:01.888. Reduction (1.7), Crossover frequency [Hz] (650), Fundamental recovery (5.0), Artifact smoothing (5.0).<br>File exported as 48kHz, 32 bit float, 2 channel (dual mono), PCM wav file. The file exported from RX 10 was opened in Cedar Cambridge for further processing.<br>Cedar Cambridge:<br>- 'NR5' noise reduction plugin, trained between 00:00:46.11 – 00:00:47:13 to obtain a noise finger print, used to reduce background noise using a notch filter.<br>Notch: -3.97 gain, 3103Hz, 0.61Q<br>- 'EQ – Precision' plugin to reduce rumble, hiss and 'boomy' speech frequencies.<br>Low shelf: -40.02 gain, 199.04Hz, 60dB/oct slope.<br>High shelf: -42.47 gain, 7556.37Hz, 54dB/oct slope.<br>Notch: -5.36dB, 224.81Hz, 0.67Q<br>-'DNS' plugin to reduce background noise. 'Low and mid' frequency range selected.<br>Level control: -28.80dB<br>Band gain controls on centre frequencies from 00:00:00 – 00:00:34.20: 27Hz (0dB), 76Hz (0), 209Hz (-1.51dB), 573Hz (-2.32dB), 1567Hz (-0.9dB), 4289Hz (0dB).<br>Band gain controls on centre frequencies from 00:00:34.20 – end of file: 27Hz (0dB), 76Hz (0dB), 209Hz (-1.51dB), 573Hz (-1.20dB), 1567Hz (-0.9dB), 4289Hz (0dB).<br>- 'FNR adaptive noise filter' plugin to reduce level of broadband noise. Resolution (high), Speed (0.20), Bias (0.0), Focus (70), Attenuation (-0.7dB).<br>- A second 'FNR adaptive noise filter' plugin to further reduce broadband noise. Resolution (optimal), Speed (0.20), Bias (0.0), Focus (70), Attenuation (-0.47dB).<br>- 'Compressor' plugin to reduce the level difference between speakers. Threshold (-21.84dB), Knee (-0.87dB), Ratio (8.78), Read ahead (6.00ms), Attack (0.27ms), Hold (25.57ms), Release (61.63ms), Makeup (18.95dB), Parallel (-50dB).<br>- 'Adaptive Limiter 2' plugin to raise the level of audio.<br>From 00:00:00 – 00:00:34.20: Gain (5), Threshold (-1), HF temporal (0), LF temporal (0), spectral (0).<br>From 00:00:34.20 – end of file: Gain (0), Threshold (-1), HF temporal (0), LF temporal (0), spectral (0).<br>All processing was peer reviewed by another competent forensic practitioner. These Cedar settings were rendered in one pass in non-real time. The rendered audio was exported from Cedar as a 24 bit (dither tpdf, no shaping), 48kHz, 2 channel (dual mono) PCM wav file.<br>To complete the normalisation the processed file was opened in SoundForge Pro v11.0 and normalised to |

 

TABLE 2

| Question 2- 1 : Enhanced Audio Examination | |
|---|---|
| **WebCode** | **Response** |
| | a peak level of -1dBFS. Normalised file saved as '23-5591_Audio_Processed.wav' (24-bit, 48 kHz, 2 channel (dual mono) PCM wav file). |
| 6D3DZ9 | Adobe Auditon 2022 with iZotope filters: -extracted and processed left channel due to slightly better signal-to-noise ratio; -applied the following filters:<br>iZotope RX 10 De-click (sensitivity 6.0, click widening 0.2)<br>notch filter (200hz, 600hz, 4000hz, 8000hz all at -50dB)<br>parametric equalizer (high pass 220hz, low pass 7500hz both at -30dB/oct)<br>hard limiter (max amplitude -3dB, input boost 18dB)<br>iZotope RX 10 Voice De-noise (adaptive, reduction 12dB)<br>iZotope RX 10 Spectral De-noise (adaptive, reduction 12 dB): -after processing normalized (-1 dB); -exported processed audio file (48kHZ, mono, 24 bit, wav) |
| 6WHM29 | Q1 was downloaded from the CTS Testing site and confirmed original vis the supplied hash value. The file was zipped so the contents was extracted, one (1) digital audio WAV file named "23-5591_Audio.wav". The file was hashed and transferred to the forensic workstation.<br>All processing was done with Adobe Audition v. 22.6.0.66<br>A. Using the Frequency Analysis tool with a sample size of 65536 and logarithmic display I identified the beeping signals as: F1 200Hz +/- 2Hz, F2 600Hz +/- 2Hz, F3 4000Hz +/- 10Hz, F4 8000Hz +/- 10Hz. Using the Marquee Selection tool and the Amplify Effect (-91 dB gain, Linked) I attenuated the digital clicks followed by the DE Clipper diagnostics (Gain Auto, Tolerance 1%, Samples 3, FFT size 128) 126 clipped areas detected and repaired.<br>B. Using the Marquee Selection tool and the Amplify Effect (-96 dB gain, Linked) I attenuated the Beeping tones found at 10 second intervals.<br>C. I found additional impact noises at approximately 45 seconds into the recording and attenuated them using the Amplify Effect (-96 dB gain, Linked). Using the Noise Reduction effect (noise reduction 47%, reduce by 16.9 dB, spectral decay 4%, precision 13, smoothing 1, transition width 0 dB) to improve signal to noise ratio.<br>D. Using the Dynamics Processing effect (customized soft knee 5-1.31:1 expansion below -76.76dB, 4-1.26:1 expansion below -40.81dB, 3-2.49:1 compression below-28.28dB, 2--4.46:1 compression below -10.27dB, -1.73:1 compression above -10.27dB) I balanced the loud and quiet talkers in the recording.<br>E. I normalized the recording to -1.0 dBFS using the Normalize effect with DC Offset at 0.0% since the Amplitude Statistics indicated no DC Bias Offset.<br>F. Derivative was saved in .WAV/PCM format at 24 bit and 48kHz sample rate. |
| 8YFKP7 | Adobe Audition was used to process the audio file. The settings have been submitted with the processed audio file. Tools include: Spot Healing Brush/Auto Heal, 20-band Graphic Equalizer, 30-band Graphic Equalizer, Noise Reduction Effect, Speech Volume Leveler Effect, and Normalize. |
| BBGFA7 | Adobe Audition was used for all processing.<br>2-A: Automatic Click Remover was used locally, with the settings tuned for each instance, supplemented by Gain Envelope.<br>2-B: Notch Filter was used to attenuate all four tones -40 dB.<br>2-C: Hiss Reduction and Parametric Equalizer were used to reduce background noise and noise outside the speech band. Automatic Click Remover and Gain Envelope were used to attenuate other clicks present.<br>2-D: Speech Volume Leveler was used to equalize near party/far party voices to a target of -15 dB RMS. |

TABLE 2

| Question 2- 1 : Enhanced Audio Examination | |
|---|---|
| **WebCode** | **Response** |
| DQ4H6Z | Used iZotope RX 9 version 9.4.0.2008 software for all the following:<br>Used spectrum analyzer on a section of audio from ~2.75 seconds to ~3.5 seconds and observed the following peaks: 200 Hz, 600 Hz, 4000 Hz and 8000 Hz<br>Used De-click: for the four clicks in the first 20 seconds, the settings were: 5.7 frequency skew, 6.2 sensitivity, 1.1 click widening (ms) on random click algorithm.<br>For clicks between 44 and 45 seconds, the frequency skew was changed to 7.1 and the other settings remained the same.<br>Used De-clip with the following settings: threshold -0.3 to -0.4, medium quality with makeup gain -3.1.<br>Used the EQ equalizer to reduce the impact of the impulses of the four bands at 200 Hz, 600 Hz, 4000 Hz and 8000 Hz<br>Used Spectral De-Noise in adaptive mode with a 3 second learn time, -0.6 threshold and 7.0 reduction, best quality, 7.0 artifact control and 3.8 smoothing.<br>Used Leveler optimized for dialogue, target level -2.0, a 6.4 responsiveness, 10 preserve dynamics and no Ess reduction or breath control.<br>Used normalize with target peak level -1.00<br>Saved file as a .wav, 24 bits, 48 kHz |
| EPR3BY | I used iZotope RX with the following filters:<br>Leveler: Used to decrease differences in level between speech. Target level: -3dB<br>Spectral Repair: To decrease amplitude of clicks in the beginning of the recording. Strength: 1.5, Surrounding region length:100%<br>EQ: To attenuate the beeps throughout the recording. Frequency bands: 200Hz, 600Hz, 4000Hz, 8000Hz, Gain: -60dB<br>Spectral De-Noise: Reduce background noise. Reduction: 18<br>Normalize: Normalize the audio to ensure it is at a reasonable level without clipping. Target peak: -1dB |
| FUF23Z | 1. Check for viruses, no viruses found. 2. Run SHA1 Hash: 5b22c920efc7d0aa9a3b3d147354d21b6f66278. 3. MediaInfo, EXIFtool, FFPROBE file forma WAV. 4. MediaInfo, EXIFtool, FFPROBE Audio File Sample Rate: 4 kHz. 5. MediaInfo, EXIFtool, FFPROBE bit depth: 2 bits. 6. Izotope/Audacity: Audio Signal found to be clipping. 7.Izotope Four Main Frequencies: 200 Hz, 600 Hz, 4,000 Hz, 8,00 Hz. 8. Izotope Review Audio (critical listening, spectrogram, spectrum analyze. 9. Izotope removed clicks. 10. Izotope applied Brickwall Filter 659 Hz, 400 Hz. 11. Izotope applied De-Noise Filtr. 12. Izotope Intermittently increased gain to low quality audio sections between 2 to 11Db. 13. Normalize audio to -1dBF. 14. Save enhanced file as: 23-5591_Audio_enhance.wav. 15. Saved File attribute: .WAV / PCM, 24bit, 48kHz |
| HLFNVV | 7) Method and tools used for audio enhancement: Using Adobe Audition 23.2.0.68, the audio waveform and spectral display of the original were reviewed. A frequency analysis was done of the tonal noise. A very narrow multinotch fiter was used 198 HZ, 561 Hz, 4001 Hz and 7970 Hz to reduce the tonal noise. The amplify tool was applied to reduce clipping. The amplify tool was also used to increase the sound of the far party speakers. The 30 band EQ tool was used to reduce noise. The normalization tool was used to normalize the entire recording to -1 db. The clarified recording was saved as a 48 kHz 24-bit stereo wav PCM file. |
| MHZKBU | on ADOBE AUDITION<br>1- NOISE REDUCTION; 2- SOUND REMOVER; 3- AMPLIFY; 4- NORMALIZE |

TABLE 2

| Question 2- 1 : Enhanced Audio Examination | |
|---|---|
| **WebCode** | **Response** |
| MWGHFV | Software used: iZotope RX 10 Advanced Audio Editor (64-bit) v10.1.0.1735<br>Audio Clarification Settings:<br>De-Clip: 0.0 dB Threshold, -1.5 dB Makeup gain, Post-limiter checked, High Quality<br>EQ: HP 111 Hz Freq, 24 dB/oct selected, 12 Hz Frequency precision<br>EQ: LP 4870 Hz Freq, 48 dB/oct selected, 12 Hz Frequency precision<br>De-Click: Time Period to Apply 0 to 1020000 samples, Single-band Algorithm, 3.0 Sensitivity, 0.0 ms Click widening<br>De-Click: Time Period to Apply 0 to 1020000 samples, Multi-band (random clicks) Algorithm, 3.9 Sensitivity, 0.9 ms Click widening, -2.0 Frequency skew<br>De-Click: Multi-band (periodic clicks) Algorithm, 2.5 Sensitivity, 0.0 ms Click widening, 1.5 Frequency skew<br>Spectral Repair:<br>Time Periods to Apply 70000 – 165658, 551608-644144, 1031431-1124685, 1511195-1604417 samples, 0-279.67 Frequency range applied to all sections, Attenuate tab, 512 Bands, Multi-resolution checked, 100% Surrounding region length, -0.6 Before/after weighting, 1.5 strength, Vertical Direction of interpolation<br>Time Periods to Apply 1991008-2084687, 2471368-2563689, 2947725-3046926, 3430000-3525478 samples, 0-279.67 Frequency range applied to all sections, Attenuate tab, 512 Bands, Multi-resolution checked, 100% Surrounding region length, -0.8 Before/after weighting, 1.5 strength, Vertical Direction of interpolation<br>Time Periods to Apply 71020-164033, 551793-643979, 1030289-1124685, 1511391-1603607 samples, Attenuate tab, 553.05-641.71 Frequency range applied to all sections, 512 Bands, Multi-resolution checked, 100% Surrounding region length, -0.3 Before/after weighting, 1.5 strength, Vertical Direction of interpolation<br>Time Periods to Apply 1992108-2079187, 2472386-2562498, 2952887-3042205, 3431855-3522768 samples, 553.05-641.71 Frequency range applied to all sections, Attenuate tab, 512 Bands, Multi-resolution checked, 74% Surrounding region length, -0.5 Before/after weighting, 1.5 strength, Vertical Direction of interpolation<br>Time Periods to Apply 70769-164761, 551099-643428, 1030000-1124095, 1511389-1603917, 1990746-2084032, 2470383-2563977, 2951359-3043677, 3430881-3524361 samples, 3875.8-4151.2 Frequency range applied to all sections, Attenuate tab, 512 Bands, Multi-resolution checked, 132% Surrounding region length, 0.5 Before/after weighting, 1.5 strength, Vertical Direction of interpolation<br>EQ: LP 7505 Hz Freq, Brickwall selected, 12 Hz Frequency precision<br>Spectral De-Noise: Time Period to Learn 1174832-1235860 samples, Parameters -0.5 Noisy and -1.0 Tonal Threshold, 11.3 Noisy and 11.3 Tonal Reduction, C Quality, 5.0 Artifact control, 5.0 Smoothing and Extreme Algorithm selected for Algorithm Behavior, 0.0 Synthesis and 10.0 Masking and 5.0 Enhancement, and 6.3 Whitening for Noise Floor, 2.5 Knee and 80 ms Release for Dynamics<br>De-Wind: 4.7 Reduction, 780 Hz Crossover frequency, 5.0 Fundamental recovery, 5.0 Artifact smoothing<br>Spectral De-Noise: Adaptive mode selected, 1.1 s Learn time, 0.0 Noisy and 0.0 Tonal Threshold, 8.3 Noisy and 8.1 Tonal Reduction, C Quality, 5.0 Artifact control, 5.0 Smoothing and Extreme Algorithm selected for Algorithm Behavior, 0.0 Synthesis and 10.0 Masking and 5.0 Enhancement, and 6.3 Whitening for Noise Floor, 2.5 Knee and 80 ms Release for Dynamics<br>Gain: Time Period to Apply 249428-250653 samples, -10.00 dB Gain<br>Leveler: Optimize for Dialogue, -18.3 Target level, 6.0 Responsiveness, 38 Preserve dynamics<br>Normalize: -1.00 dFBS Target peak level<br>Export: Wav, 24-bit Bit depth, None (truncate) Dither<br>For further information, please refer to the document 23-5591 Audio Clarification Documentation |

TABLE 2

| Question 2- 1 : Enhanced Audio Examination | |
|---|---|
| **WebCode** | **Response** |
| Q2JNJQ | Processing in Adobe Audition:<br>- Channel separation and removal of the right channel, due to it having more pre-processing. Worked from the left channel. - DeClipper (restore slightly trimmed). - DeClicker (medium reduction). - Frequency reduction from 8000 Hz onwards, to eliminate clipping artifacts, FFT filter. - Band rejection filter for frequencies 200 Hz, 600 Hz, 4000 Hz and 8000 Hz to remove frequencies (super narrow width) at specific times when the beep sound is heard. - Manual reduction and cleaning of other artifacts. - Noise reduction 5 dB and 25%, 4096 FFT size. - Reverb reduction to 20%. - Voice leveling with single band compressor (-10 dB threshold). - Voice volume leveler by 5% (-18 dB RMS and 45 dB). - Normalize at -1 dB.<br>The audio is saved in .wav (PCM) format, 2 channels, 24 bit depth, 48000 Hz sample rate. |
| RVNFDN | I used Adobe Audition version 2023 to examine and process the audio as requested.<br>I extracted out the Left channel to use for the enhanced audio since it appears to have more data in the higher frequencies.<br>I used markers to establish selections for the following areas: • the 8 "beeps" that can be heard throughout the file (~2 seconds each); • the 4 "clicks" in the first 20 seconds (less than 0.1 seconds each); • the 4 "pulses" that can be heard starting at approximately 44 seconds from the start (less than 0.1 seconds each).<br>I used the Parametric EQ filter to create a bandpass filter to reduce frequencies outside of 50 Hz and 6000 Hz.<br>I selected each of the four clicks and four pulses separately and used the Auto Heal Selection tool to reduce the amplitude of the clicks and pulses in these areas.<br>I selected each of the eight beeps separately and reduced the noise in these areas using the Effects Rack with 3 Parametric EQ filters to create notch filters to reduce the frequencies at 200 Hz, 600 Hz, 4000 Hz, and 8000 Hz.<br>I selected each of the eight beeps separately and used the Parametric EQ filter to create notch filter at 200 Hz to reduce the tones still present in this area.<br>I used the Effects Rack to process the entire recording after running the processes above.<br>1. Hard Limitr: a.limited amplitude to -1dB, no increase in amplitude<br>2. iZotope RX 10 Spectral De-noise filter as a VSPlugin: a. Adaptive mode, Quality C, -30 dB Reduction with 3.0 second learn time<br>3. Hard Limitr: a.limited amplitude to -9dB, 32 dB boost<br>4. Parametc EQ: a. bandpass between 100 Hz and 6000 z; b. notches at 200 Hz, 400 Hz, and 4000 Hz<br>I selected each of the four pulses separately again and the Parametric EQ to create a 300 Hz High Pass filter to reduce the noise present in this area.<br>I ran the Normalize filter to increase the peak amplitude to -1dB. |

TABLE 2

| Question 2- 1 : Enhanced Audio Examination | |
|---|---|
| **WebCode** | **Response** |
| V2M4JK | To address the perceived clipped audio the entire signal was processed with De-clip with setting: -1 in the software Izotope RX 9.<br>To address the four digital clicks in the first 20 s of the file, the first 20 s were processed with De-click in the same software. Settings used were: Algorithm: Single-band, Sensitivity: 4.2, Click widening [ms]: 0.<br>A second pass with De-click was done over the click at 05 s and 200 ms to further reduce it, settings used were: Algorithm: Multi-band (Random clicks), Sensitivity: 9.6, Click widening [ms]: 1.5.<br>To reduce the beep that occurred at every 10th second the De-hum module in Izotope RX 9 was used. In order to adapt the filter to the specific signal, the beep at 11 s 453 ms was selected and used for training. Each instance of the beep was individually selected and filtered. The settings used in the De-hum module were: Sensitivity: 8.4; Band: 370; Filter Q: 240.<br>The resulting file was then opened in Wavelab for further processing. To reduce the level difference in the speech and between the voices the Waves MV2 compressor/limiter plug-in was used with the settings: Low Level: 28, High Level: -19, Output: -9.7.<br>After this step a 10 band EQ was utilized to reduce the impact of the background sound on the voices as well as boosting frequencies in order to make the voice more intelligible.<br>The EQ was a Waves Q10 plug-in with the settings:<br>Three High-pass filters at 207 Hz Q: 7.0 Gain: 0<br>Peak-filter at 351 Hz Q: 7.0 Gain: -0.8<br>Peak-filter at 578 Hz Q: 7.0 Gain: 4.5<br>Peak-filter at 1189 Hz Q: 7.0 Gain: 4.5<br>Peak-filter at 2089 Hz Q: 7.0 Gain: 1.3<br>Peak-filter at 3943 Hz Q: 7.0 Gain: 0.2<br>This processed audio was rendered out from Wavelab and opened in Izotope RX 9 for additional processing to reduce the impact of the background sound on the speech and voices. The module Dialogue Isolate was used with the settings: Dialogue Gain: 0, Noise gain: -12, Sensitivity: 5.0, Ambience preservation: 50, Quality: Best.<br>This file was saved and once again opened in Wavelab, this time for dynamic compression in order to further reduce the level differences in the speech and voices. The Waves MV2 plug-in was used with the settings: Low level: 16.4, High Level: -14.7, Output: -3.<br>As a last step the signal was normalized to -1 dBFS with the function Level Normalizer in Wavelab. The settings were: -1 dB, Reference: Digital Peaks<br>The resulting file was saved as a Wave(PCM), 24 bit, fs. 48 kHz, stereo file. |

TABLE 2

| Question 2- 1 : Enhanced Audio Examination | |
|---|---|
| **WebCode** | **Response** |
| WAWTWJ | (08/09/2023)[Initials]<br>Downloaded and opened zip file 23-5591_Audio.zip and extracted zip file to Documents folder on corporate laptop.<br>Ran jacksum on 23-5591_Audio.wav to generate SHA-1:<br>5b22c920efc7d0aa9a3b3d147354d21b6f66278a.<br>Copied extracted folder to scientific server folder 25875_[Initials]_PT<br>Ran jacksum of server copy of 23-5591_Audio.wav to generate SHA-1:<br>5b22c920efc7d0aa9a3b3d147354d21b6f66278a.<br>Ran FFPROBE on 23-5591_Audio.wav to check properties.<br>File is 1m21.51s duration, 48kHz sample rate, 2 channels, 24-bit resolution.<br>(08/09/2023)[Initials]<br>Opened 23-5591_Audio.wav in IZotope RX9.<br>Audio appears to be a body worn camera audio recording between police member, offender and other parties. The audio is marred by broadband background noise, periodic camera beeps, digital clicks and ocassional clipping. The speech levels between parties are uneven with the voice of the camera wearer predominant, and other parties lower in level. Adaptive noise reduction, EQ, de-clip, de-clicking and dynamics control may be able to rectify issues and produce a more balanced product with wider signal to noise ratio.<br>Ran Waveform Stats and showed True Peak Levels at +2.71dB (L) and +2.23dB (R)<br>Ran Spectrum and showed frequencies of BWC beeps to be 200Hz, 600Hz, 4kHz, 8kHz.<br>Applied EQ as band pass filter:<br>LPF Brickwall @ 7kHz<br>HPF Brickwall @ 250Hz.<br>This removed unwanted low and high level noise as well as beep frequencies at 200Hz and 8kHz.<br>Applied EQ as notch filter:<br>50dB cut 37.1(Q) at 600Hz<br>40dB cut 35.1(Q) at 4Kz<br>Applied De-Click to attenuate digital clicks at region between 5sec and 20sec mark<br>Applied Clip Gain to selected regions to amplify lower level speech to balance against higher level speech.<br>Applied Spectral De-Noise (Adaptive mode) on amplified regions to match noise levels of louder regions.<br>Applied EQ with 3dB boost at 2500Hz to accentuate speech frequencies at high mids.<br>Applied Compressor/Limiter to reduce overall dynamic range and limit headroom to -8dB.<br>Applied Normalisation to -1dBFS.<br>(11/09/2023)[Initials]<br>File saved as 23-5591_Audio.wav at 48kHz 24-bit PCM.<br>(Screenshots for settings and XML report available if requested).<br>Overall signal to noise ratio between speech and background noise was increased. Speech at the first 35secs was low in level and appeared to be a distance from the recording device and marred by background noise. Levels in that section were increased with no significant improvement to intelligibility. Remainder of recording had slight improvements to the intelligibility of person of interest. |
| XMZB9J | iZotope RX10: De-click, De-clip, Spectral repair, Dialogue isolate, Normalize |

TABLE 2

| Question 2- 1 : Enhanced Audio Examination | |
|---|---|
| **WebCode** | **Response** |
| YZKD4F | Examined and processed using Cedar Cambridge v14<br>Selected clicks and applied 'Manual Declick - model B'<br>4 x clicks between 5s and 20s; 4 additional clicks at approx 44s.<br>Passed Declicked signal through the following processing chain:<br>Declip > EQ-P (IIR) > Phase Correction > Mono Sum > EQ-L (FIR) > Compressor > Broadband Noise reduction (FNR) > Rendered WAV result > Peak Normaliser > Final WAV Result.<br>Module settings:<br>DeClip - Clip thresholds manually set to -/+80%FS - captured additional clipping distortion that was too long / complex for manual DeClick.<br>EQ-P (attenuated tones)<br>200Hz Q60 -50dB; 600Hz Q100 -40dB; 4000Hz Q80 -50dB; 8000Hz Q80 -50dB.<br>Phase correction: Rch +150 microseconds<br>Mono Sum - combined Ch 1 and 2. Balance adjusted to prefer Ch1<br>EQ-L (Linear Phase, spectral shaping to improve S/N & subjective intelligibility)<br>Low shelf 220Hz -20dB 18db/8ve<br>750Hz -6dB Q=2; 420Hz -7dB Q=9; 3200Hz +5dB Q=0.6; 2700Hz -6dB Q=16<br>High Shelf: 7500Hz -6dB 12db/8ve<br>Compressor: threshold -20dB; ratio 10:1; knee: 2dB; attack: 0.1ms; release 50ms; post-gain: 12dB<br>FNR (broadband NR): Adaption speed = 0.8Hz; Adaption resolution: 'Optimal'; attenuation: -2dB<br>Peak Normalise: -1dB<br>Hi Shelf 7500Hz -8dB 12dB/8ve |

Question 2-1: Note the methods or tools used and the settings for the audio enhancement here.

Consensus Result: This was a free form question on methods and tools used. No consenus response expected.

# Forensic Audio Enhancement Observations

TABLE 3

**Part 2: Enhanced Audio Instructions** - Use any methods or software tools deemed necessary to process the file, remove noise, or otherwise improve the intelligibility of the clip. The derivative file should achieve the following goals: Decrease the dynamic range of the multiple speakers, Increase the intelligibility of the words spoken, Decrease the level of noise and other unwanted sounds and Be free of clipping and other excessive artifacts or distractions from over processing.

**A.** Address / repair / reduce the impact of the four digital clicks / impulses in the first 20 seconds of the audio file, and any other clipping or distortions.

**B.** Using EQ or some other frequency specific tool, reduce or remove the beeps that repeat every 10 seconds. Be careful not to overprocess or affect more of the frequency spectrum than necessary.

**C.** Improve the signal to noise ratio by increasing the difference in level between the speaking and background sounds.

**D.** Reduce the difference in level between the loud and quiet parts of the speech. (Each speaker should be heard at a comparable level to each other if possible.)

**E.** Normalize or manually adjust the overall level of the clip to achieve a peak level value of -1dBFS.

**F.** Save your derivative file with the following parameters: a) Format: .wav/PCM, b) 24 bit, 48,000 Hz sample rate.

| WebCode | Observational Notes |
|---|---|
| 2PMJJE | Submitted enhanced file(s) were reviewed by an expert who confirmed that this participant completed all requested audio enhancement steps with the following exception(s). Step C observational note: the relative level of the background sound has stayed about the same as in the original audio file. |
| 3XW9WD | Submitted enhanced file(s) were reviewed by an expert who confirmed that this participant completed all requested audio enhancement steps with the following exception(s). Step D observational note: the relative level of the different speakers remains about the same as in the original audio file. |
| 62ARUD | Submitted enhanced file(s) were reviewed by an expert who confirmed that this participant completed all requested audio enhancement steps. |
| 69VK7C | Submitted enhanced file(s) were reviewed by an expert who confirmed that this participant completed all requested audio enhancement steps with the following exception(s). Step C observational note: the relative level of the background sound has stayed about the same as in the original audio file. |
| 6D3DZ9 | Submitted enhanced file(s) were reviewed by an expert who confirmed that this participant completed all requested audio enhancement steps. |
| 6WHM29 | Submitted enhanced file(s) were reviewed by an expert who confirmed that this participant completed all requested audio enhancement steps. |
| 8YFKP7 | Submitted enhanced file(s) were reviewed by an expert who confirmed that this participant completed all requested audio enhancement steps. |
| BBGFA7 | Submitted enhanced file(s) were reviewed by an expert who confirmed that this participant completed all requested audio enhancement steps. |
| DQ4H6Z | Submitted enhanced file(s) were reviewed by an expert who confirmed that this participant completed all requested audio enhancement steps. |
| EPR3BY | Submitted enhanced file(s) were reviewed by an expert who confirmed that this participant completed all requested audio enhancement steps with the following exception(s). Step E observational note: peak level is at -3 dBFS. |
| FUF23Z | Submitted enhanced file(s) were reviewed by an expert who confirmed that this participant completed all requested audio enhancement steps. |
| HLFNVV | Submitted enhanced file(s) were reviewed by an expert who confirmed that this participant completed all requested audio enhancement steps with the following exception(s). Step C observational note: the level of dominant speech has been lowered and the level of background sound has stayed relatively the same. Step E observational note: peak level is at -3.83 dBFS. |

TABLE 3

**Forensic Audio Enhancement Observations**

| WebCode | Observational Notes |
|---|---|
| MHZKBU | Submitted enhanced file(s) were reviewed by an expert who confirmed that this participant completed all requested audio enhancement steps with the following exception(s). Step D observational note: the relative level of the different speakers remains about the same as in the original audio file. Step E observational note: peak level is at -.12 dBFS. |
| MWGHFV | Submitted enhanced file(s) were reviewed by an expert who confirmed that this participant completed all requested audio enhancement steps. |
| Q2JNJQ | Submitted enhanced file(s) were reviewed by an expert who confirmed that this participant completed all requested audio enhancement steps. |
| RVNFDN | Submitted enhanced file(s) were reviewed by an expert who confirmed that this participant completed all requested audio enhancement steps. |
| V2M4JK | Submitted enhanced file(s) were reviewed by an expert who confirmed that this participant completed all requested audio enhancement steps with the following exception(s). Step C observational note: the level of background sound has increased slightly. |
| WAWTWJ | Submitted enhanced file(s) were reviewed by an expert who confirmed that this participant completed all requested audio enhancement steps with the following exception(s). Step C observational note: the level of dominant speech has been lowered and the level of background sound has stayed relatively the same. |
| XMZB9J | Submitted enhanced file(s) were reviewed by an expert who confirmed that this participant completed all requested audio enhancement steps. |
| YZKD4F | Submitted enhanced file(s) were reviewed by an expert who confirmed that this participant completed all requested audio enhancement steps with the following exception(s). Step C observational note: the level of dominant speech has been lowered and the level of background sound has stayed relatively the same. |

# Additional Comments

TABLE 4

| WebCode | Additional Comments |
|---|---|
| 3XW9WD | It's challenging to distinguish between background noise and a person's soft voice during editing, and it's also difficult to equalize the volume of all individuals when two or more people are conversing. |
| 62ARUD | Since October of 2018 the [Laboratory] (Digital Forensic Analysis Section) no longer considers audio enhancement to be a forensic process. |
| 6WHM29 | The FFT tool in Adobe Audition did not have the resolution to pinpoint the exact frequency of the beeping noise in the recording so a range was given.<br>Since the conversation was targeted in the request the derivative could have been down sampled rather than keeping the derivative at 48kHz sampling rate. Reducing the bandwidth would also help reducing the beeps and clicks. |
| 8YFKP7 | The following question is extremely confusing. By not including any units of measurement or times, it is difficult to understand if you are looking four the first four beeps or the first four sets of four beeps and if you are asking for the overall range or most common frequencies of the beeps. I answered with most common frequency spikes in the four-beep set.<br>1-6). Starting with the lowest frequency (i.e. Frequency 1) and moving upward, what are the four main frequencies (or frequency ranges) that make up the beeps / audio prompts that repeat every 10 seconds?<br>The processing directions are far beyond what would be done for an audio clarification. The voices were low but intelligible without additional processing. |
| WAWTWJ | XML reports and screenshots available on request. |
| YZKD4F | Noted difference between channels. Channel 2 (right) appeared to be bandlimited and pre-processed. We decided that result should be presented in mono to ensure optimised intelligibility on multi-speaker systems (eg: courtooms with ceiling mounted stereo systems). 2 solutions attempted Option 1: discard Rch and retain full bandwidth channel only; Option 2: correct phase between channels and combine to mono. Option 2 was used in the final result. |

-End of Report-
(Appendix may follow)

**Collaborative Testing Services ~ Forensic Testing Program**

# Test No. 23-5591: Forensic Audio Analysis

DATA MUST BE SUBMITTED BY **Sept. 18, 2023, 11:59 p.m. EDT** TO BE INCLUDED IN THE REPORT

Participant Code: U1234H WebCode: KMWPQU

**Scenario:**

An audio file has been submitted to you for examination. You are asked to minimize the distracting elements, and clarify the speech from the incident contained in the file, so that the subjects can be better heard and understood.

*This test is designed to measure your knowledge and skill in the following digital forensic audio processes: Data verification, Media characterization, Data analysis, Signal analysis and Enhancement.*

*The skills assessed in this exercise are based on the following best practice documents from the Scientific Working Group for Digital Evidence (swgde.org):*

*1. Forensic Audio Core Competencies*
*2. Best Practices for the Enhancement of Digital Audio*
*3. Best Practices for Forensic Audio*

*Because of the inherent subjectivity of audio "enhancement" due to differences in an individual's personal preferences and hearing, you will be asked to perform specific tasks including identifying individual elements of the audio signal, taking measurements, and processing the file in a way that is designed to show your understanding of a certain principle or concept. It is critical that you read the instructions carefully and execute the tasks in order. Be aware that the process chain you use may affect your ability to reach the target response.*

**Evidence:**

To verify a complete and accurate download, use the tool of your choice to verify the integrity of the file.
23-5591_Audio.zip MD5 hash value: d6b7d5a4b2e44e31c0c4a833726df1c8
23-5591_Audio.zip SHA1 hash value: d1c39bb4ec2407d3d5271c43b17e3cee5246c840

Participant Code: U1234H
WebCode: KMWPQU

1-1). What is the SHA1 hash value of the audio file?

1-2). What is the file format of the audio file?

1-3). What is the sample rate of the audio file?

1-4). What is the bit depth of the audio file?

1-5). Is the audio signal clipping?

1-6). Starting with the lowest frequency (i.e. Frequency 1) and moving upward, what are the four main frequencies (or frequency ranges) that make up the beeps / audio prompts that repeat every 10 seconds?

Frequency 1

Frequency 2

Frequency 3

Frequency 4

## Part 2: Enhanced Audio Instructions

Use any methods or software tools deemed necessary to process the file, remove noise, or otherwise improve the intelligibility of the clip. The derivative file should achieve the following goals: Decrease the dynamic range of the multiple speakers, Increase the intelligibility of the words spoken, Decrease the level of noise and other unwanted sounds and Be free of clipping and other excessive artifacts or distractions from over processing.

A. Address / repair / reduce the impact of the four digital clicks / impulses in the first 20 seconds of the audio file, and any other clipping or distortions.

B. Using EQ or some other frequency specific tool, reduce or remove the beeps that repeat every 10 seconds. Be careful not to overprocess or affect more of the frequency spectrum than necessary.

C. Improve the signal to noise ratio by increasing the difference in level between the speaking and background sounds.

D. Reduce the difference in level between the loud and quiet parts of the speech. (Each speaker should be heard at a comparable level to each other if possible.)

**ONLY AFTER the above processing has been completed perform the following tasks:**

E. Normalize or manually adjust the overall level of the clip to achieve a peak level value of -1dBFS.

F. Save your derivative file with the following parameters:
a. Format: .wav / PCM
b. 24 bit, 48,000 Hz Sample rate.

Uploaded file name:

2-1). Note the methods or tools used and the settings for the audio enhancement here.

## Additional Comments

***Please note:*** *Any additional formatting applied in the free form space below will not transfer to the Summary Report and may cause your information to be illegible. This includes additional spacing and returns that present your responses in lists and tabular formats.*

# RELEASE OF DATA TO ACCREDITATION BODIES

**The Accreditation Release is accessed by pressing the "Continue to Final Submission" button online and can be completed at any time prior to submission to CTS.**

**CTS submits external proficiency test data directly to ASCLD/LAB, ANAB, and/or A2LA. Please select one of the following statements to ensure your data is handled appropriately.**

○ This participant's data is intended for submission to ASCLD/LAB, ANAB, and/or A2LA. (Accreditation Release section below must be completed.)

○ This participant's data is **not** intended for submission to ASCLD/LAB, ANAB, and/or A2LA.

Have the laboratory's designated individual complete the following steps
**only if your laboratory is accredited in this testing/calibration discipline**
by one or more of the following Accreditation Bodies.

**Step 1: Provide the applicable Accreditation Certificate Number(s) for your laboratory**

**ANAB** Certificate No.
(Include ASCLD/LAB Certificate here)

**A2LA** Certificate No.

**Step 2: Complete the Laboratory Identifying Information in its entirety**

Authorized Contact Person and Title

Laboratory Name

Location (City/State)