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FORENSIC TESTING PROGRAM

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Forensic Audio Analysis
Test No. 24-5591 Summary Report

Participants were provided with an audio evidence file and asked to examine it using their own tools and methods. Data were returned from 31 participants and are compiled in the following tables:

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 digital audio recording of a microcassette tape recording was executed.

The .wma 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.

VERIFICATION: Predistribution results were consistent with each other and the manufacturer's preparation information. 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: QuickHash-GUI (version 3.3.4), MediaInfo (version 24.04), iZotope RX 11 Advanced Audio Editor, and Audacity (version 3.5.1). CTS does not endorse any particular tools.

PLEASE NOTE: Questions marked with asterisks (**) did not show a clear consensus during preliminary review of the participants' responses. Further information and discussion will be available in the final summary report.

SCENARIO PROVIDED TO PARTICIPANTS

A microcassette recording was made of an attorney calling in to a disciplinary hearing. Later, the microcassette was digitized into a .wma file (“01 Track 1.wma”). The submitter of this recording stated that the original microcassette is no longer available, but they need the speech in the digitized copy clarified in order to have it accurately transcribed.

Manufacturer's Information, continued

Question **Manufacturer's Response - Examination Questions**

1-1 What is the SHA-256 hash value of the 01 Track 1.wma file?

Manufacturer's Response:

58A3049158D544B9D0D4661749B6A53DB2E010B6D9C57FA2BA03220F4B232F94

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

Manufacturer's Response:

Windows Media Audio / Windows Media / WMA

1-3 How many audio channels are contained in the audio file? Report a numeric value.

Manufacturer's Response:

2

1-4 What is the length in time of the audio file?

Manufacturer's Response:

10 minutes 7 seconds / 10:07

1-5 What is the bit depth of the audio file?

Manufacturer's Response:

16 bit

1-6 What is the sample rate of the audio file?

Manufacturer's Response:

44.1 kHz / 44,100 Hz

1-7** What is the average value of the fundamental frequency of the dominant tonal noise present? Are harmonics of this tone present?

Manufacturer's Response:

Fundamental frequency average value: 179.8 Hz / 180 Hz
Harmonics present? Yes

Question **Manufacturer's Response - Enhanced Audio Examination**

2-1 Describe the workflow used to clarify the audio, including specific software and filters applied.

Manufacturer's Response:

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

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. Refer to Manufacturer's Information for preparation details.

A total of 31 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 but two examination questions achieved consensus responses greater than 95%. Question 1-5 did achieve a consensus, where 77.4% of participants reported the Manufacturer's response of 16 bits, and 16% reported 32 bits. Question 1-7 was a two-part question. For this report, since a consensus was achieved for one part of the question and not the other, Question 1-7 is being shown together and as two individual questions. Separated, the questions are identified as 1-8 and 1-9. For question 1-8, which asks for the average value of the fundamental frequency of the dominant tonal noise present in the audio file, only 67% of participants reported the Manufacturer's response of 179 or 180Hz. For question 1-9, which asks whether harmonics are present in this tone, 96.8% of participants reported the Manufacturer's response that they were present.

In a separate section of this test, participants were asked 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 the observational notes present in Table 3.

Forensic Audio Examination Responses

TABLE 1

Question 1- 1 : Examination Questions

Question 1-1: What is the SHA-256 hash value of the 01 Track 1.wma file?

Manufacturer's Response: 58A3049158D544B9D0D4661749B6A53DB2E010B6D9C57FA2BA03220F4B232F94

WebCode	Response
38J7PA	58a3049158d544b9d0d4661749b6a53db2e010b6d9c57fa2ba03220f4b232f94
44DZM9	58A3049158D544B9D0D4661749B6A53DB2E010B6D9C57FA2BA03220F4B232F94
44HGK8	58A3049158D544B9D0D4661749B6A53DB2E010B6D9C57FA2BA03220F4B232F94
73WPM8	58a3049158d544b9d0d4661749b6a53db2e010b6d9c57fa2ba03220f4b232f94
7VUMEZ	58a3049158d544b9d0d4661749b6a53db2e010b6d9c57fa2ba03220f4b232f94
937E24	58a3049158d544b9d0d4661749b6a53db2e010b6d9c57fa2ba03220f4b232f94
9KQ6Z3	58A3049158D544B9D0D4661749B6A53DB2E010B6D9C57FA2BA03220F4B232F94
9LH8KX	58A3049158D544B9D0D4661749B6A53DB2E010B6D9C57FA2BA03220F4B232F94
9LLM26	58a3049158d544b9d0d4661749b6a53db2e010b6d9c57fa2ba03220f4b232f94
AD27MZ	1.1 58A3049158D544B9D0D4661749B6A53DB2E010B6D9C57FA2BA0322F4B232F94
APE4R4	58a3049158d544b9d0d4661749b6a53db2e010b6d9c57fa2ba03220f4b232f94
AZWRHV	58a3049158d544b9d0d4661749b6a53db2e010b6d9c57fa2ba03220f4b232f94
BU6ZTY	58a3049158d544b9d0d4661749b6a53db2e010b6d9c57fa2ba03220f4b232f94
CVDERT	58a3049158d544b9d0d4661749b6a53db2e010b6d9c57fa2ba03220f4b232f94
FVJZUP	58A3049158D544B9D0D4661749B6A53DB2E010B6D9C57FA2BA03220F4B232F94
GFQLDW	58a3049158d544b9d0d4661749b6a53db2e010b6d9c57fa2ba03220f4b232f94
GNZAQV	58a3049158d544b9d0d4661749b6a53db2e010b6d9c57fa2ba03220f4b232f94
HU2LMR	58A3049158D544B9D0D4661749B6A53DB2E010B6D9C57FA2BA03220F4B232F94
HX2XJR	58a3049158d544b9d0d4661749b6a53db2e010b6d9c57fa2ba03220f4b232f94
J3UD9Q	58a3049158d544b9d0d4661749b6a53db2e010b6d9c57fa2ba03220f4b232f94
LAN7UQ	58a3049158d544b9d0d4661749b6a53db2e010b6d9c57fa2ba03220f4b232f94
N79P6G	58a3049158d544b9d0d4661749b6a53db2e010b6d9c57fa2ba03220f4b232f94
N7GLJM	58A3049158D544B9D0D4661749B6A53DB2E010B6D9C57FA2BA03220F4B232F94
PDQBXL	58a3049158d544b9d0d4661749b6a53db2e010b6d9c57fa2ba03220f4b232f94
T94UVJ	58a3049158d544b9d0d4661749b6a53db2e010b6d9c57fa2ba03220f4b232f94
UAUK9B	58a3049158d544b9d0d4661749b6a53db2e010b6d9c57fa2ba03220f4b232f94
UP9QJB	58A3049158D544B9D0D4661749B6A53DB2E010B6D9C57FA2BA03220F4B232F94

TABLE 1

Question 1- 1 : Examination Questions	
WebCode	**Response**
X4F2LE	58A3049158D544B9D0D4661749B6A53DB2E010B6D9C57FA2BA03220F4B232F94
YPVLHE	58a3049158d544b9d0d4661749b6a53db2e010b6d9c57fa2ba03220f4b232f94
Z2HGN6	58A3049158D544B9D0D4661749B6A53DB2E010B6D9C57FA2BA03220F4B232F94
ZUY4TC	58a3049158d544b9d0d4661749b6a53db2e010b6d9c57fa2ba03220f4b232f94

Question 1-1: What is the SHA-256 hash value of the 01 Track 1.wma file?

Consensus Result: 58A3049158D544B9D0D4661749B6A53DB2E010B6D9C57FA2BA03220F4B232F94

TABLE 1

Question 1- 2 : Examination Questions

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

Manufacturer's Response: Windows Media Audio / Windows Media / WMA

WebCode	Response
38J7PA	Advanced Systems Format (ASF) file container, containing Windows Media Audio 9.2 constant bit rate audio data
44DZM9	Windows Media Audio or .WMA
44HGK8	Windows Media (.wma)
73WPM8	.WMA
7VUMEZ	WMA (Windows Media)
937E24	WMA
9KQ6Z3	WMA
9LH8KX	wma Windows Media Audio
9LLM26	WMA V2
AD27MZ	WMA – Windows Media Audio
APE4R4	Windows Media
AZWRHV	WMA (Windows Media Audio)
BU6ZTY	.wma (Windows Media Audio)
CVDERT	windows Media Audio
FVJZUP	wma
GFQLDW	Windows Media Audio (wma)
GNZAQV	WMA
HU2LMR	wma
HX2XJR	windows media audio(WMA) Version 2
J3UD9Q	Window Media Audio
LAN7UQ	Windows Media Audio (.wma)
N79P6G	WMA
N7GLJM	WMA
PDQBXL	WMAudioV2
T94UVJ	Windows Media Audio
UAUK9B	Format of file is WMA version 2. File container is asf (advanced system format).
UP9QJB	.wma
X4F2LE	.wma

TABLE 1

Question 1- 2 : Examination Questions	
WebCode	**Response**
YPVLHE	WMA (Windows Media Audio)
Z2HGN6	WMA (Version 2)
ZUY4TC	Windows Media Audio(wma)

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

Consensus Result: Windows Media Audio / Windows Media / WMA

TABLE 1

Question 1- 3 : Examination Questions

Question 1-3: How many audio channels are contained in the audio file? Report a numeric value.

Manufacturer's Response: 2

WebCode	Response
38J7PA	2
44DZM9	2
44HGK8	2
73WPM8	2
7VUMEZ	2
937E24	2
9KQ6Z3	2
9LH8KX	2
9LLM26	2
AD27MZ	2
APE4R4	2
AZWRHV	2
BU6ZTY	2
CVDERT	2
FVJZUP	2
GFQLDW	2
GNZAQV	2
HU2LMR	2
HX2XJR	2
J3UD9Q	2
LAN7UQ	2
N79P6G	2
N7GLJM	2
PDQBXL	2
T94UVJ	2
UAUK9B	02
UP9QJB	2
X4F2LE	2

TABLE 1

Question 1- 3 : Examination Questions	
WebCode	**Response**
YPVLHE	2
Z2HGN6	2
ZUY4TC	2

Question 1-3: How many audio channels are contained in the audio file? Report a numeric value.

Consensus Result: 2

TABLE 1

Question 1- 4 : Examination Questions

Question 1-4: What is the length in time of the audio file?

Manufacturer's Response: 10 minutes 7 seconds / 10:07

WebCode	Response
38J7PA	10 minutes, 7.062 seconds
44DZM9	00:10:07.06
44HGK8	10:07 minutes (10 minutes + 7 seconds)
73WPM8	00:10:07
7VUMEZ	00:10:07.063 (h:m:s:.ms)
937E24	10 min, 7 sec
9KQ6Z3	10 min 7 s
9LH8KX	10:07.062 10 minutes, 7 seconds, 62 milliseconds.
9LLM26	10 min 7 s
AD27MZ	10 minutes 7 seconds
APE4R4	10:07.062
AZWRHV	10min 7s 62ms
BU6ZTY	10 minutes 7 seconds (00:10:07.062)
CVDERT	(00:10:07) ten minutes and seven seconds.
FVJZUP	10 minutes 7 seconds
GFQLDW	00:10:07:062 (hh:mm:ss:ms)
GNZAQV	10 min 7 s
HU2LMR	10 min 7 sec
HX2XJR	10 min 07 sec
J3UD9Q	10 minutes 7 seconds
LAN7UQ	10 min 7 seconds
N79P6G	9793895
N7GLJM	10 min 7 sec
PDQBXL	10:07.062
T94UVJ	around 10 min 7 sek.
UAUK9B	10 minutes 7 seconds.
UP9QJB	10 minutes 7 seconds 62 milliseconds
X4F2LE	10:07:062

TABLE 1

Question 1- 4 : Examination Questions	
WebCode	**Response**
YPVLHE	10:07.062
Z2HGN6	10:07.062
ZUY4TC	00:10:07.108(10 minutes and 07.108 seconds)

Question 1-4: What is the length in time of the audio file?

Consensus Result: 10 minutes 7 seconds / 10:07

TABLE 1

Question 1- 5 : Examination Questions

Question 1-5: What is the bit depth of the audio file?

Manufacturer's Response: 16 bit

WebCode	Response
38J7PA	32 bits, but this may be a player representation after conversion/extraction of the WMA-data; the WMA codec standard from Microsoft states an encoder bit depth of 16 bits
44DZM9	16
44HGK8	16 Bits
73WPM8	16 bits
7VUMEZ	16
937E24	16 bits
9KQ6Z3	16
9LH8KX	32-bit
9LLM26	16 bits
AD27MZ	16 bits
APE4R4	16 bits
AZWRHV	16 bit
BU6ZTY	16 bit
CVDERT	32 bits
FVJZUP	16 bits
GFQLDW	32 bits float
GNZAQV	16
HU2LMR	16 bit
HX2XJR	16 bits
J3UD9Q	16 bits
LAN7UQ	16
N79P6G	129 kbps
N7GLJM	32 bit
PDQBXL	16-bit
T94UVJ	16
UAUK9B	16 bits
UP9QJB	16 Bit
X4F2LE	16 bits

TABLE 1

Question 1- 5 : Examination Questions	
WebCode	**Response**
YPVLHE	16
Z2HGN6	128 kb/s
ZUY4TC	32 bits

Question 1-5: What is the bit depth of the audio file?

Consensus Result: 16 bit

TABLE 1

Question 1- 6 : Examination Questions

Question 1-6: What is the sample rate of the audio file?

Manufacturer's Response: 44.1 kHz / 44,100 Hz

WebCode	Response
38J7PA	44100 Hz
44DZM9	44.1 kHz
44HGK8	44.1 KHz
73WPM8	44.1 kHz
7VUMEZ	44 100 Hz
937E24	44.1 kHz
9KQ6Z3	44.1 kHz
9LH8KX	44100 Hz
9LLM26	44.1 kHz
AD27MZ	44.1 kHz
APE4R4	44.1 kHz
AZWRHV	44.1 kHz
BU6ZTY	44.1 kHz
CVDERT	44100 Hz
FVJZUP	44.1 kHz
GFQLDW	44100 Hz
GNZAQV	44100 Hz
HU2LMR	44.1 kHz
HX2XJR	44.1 kHz
J3UD9Q	44.1 kHz
LAN7UQ	44.1 kHz
N79P6G	44100
N7GLJM	44.1 Khz
PDQBXL	44,100 Hz
T94UVJ	44100
UAUK9B	44.1 kHz
UP9QJB	44.1 kHz
X4F2LE	44.1 kHz

TABLE 1

Question 1- 6 : Examination Questions	
WebCode	**Response**
YPVLHE	44.1 kHz
Z2HGN6	44,1 kHz
ZUY4TC	44100 Hz

Question 1-6: What is the sample rate of the audio file?

Consensus Result: 44.1 kHz / 44,100 Hz

TABLE 1

Question 1- 7 : Examination Questions

Question 1-7: What is the average value of the fundamental frequency of the dominant tonal noise present? Are harmonics of this tone present?

Manufacturer's Response: Fundamental frequency average value: 179.8 Hz / 180 Hz
Harmonics present? Yes

WebCode	Response ** Inconsistencies not highlighted; No consensus achieved **
38J7PA	Fundamental frequency average value: 180 Hz, Harmonics present?: Yes
44DZM9	Fundamental frequency average value: Fundamental frequency average value: 60 Hz, Harmonics present?: Yes
44HGK8	Fundamental frequency average value: Harmonics present?: Yes
73WPM8	Fundamental frequency average value: Mean F0 = 170 Hz, Harmonics present?: Yes
7VUMEZ	Fundamental frequency average value: 179,215 Hz, Harmonics present?: Yes
937E24	Fundamental frequency average value: 179.8 Hz, Harmonics present?: Yes
9KQ6Z3	Fundamental frequency average value: 180, Harmonics present?: Yes
9LH8KX	Fundamental frequency average value: 180Hz, Harmonics present?: Yes
9LLM26	Fundamental frequency average value: 180 Hz, Harmonics present?: Yes
AD27MZ	Fundamental frequency average value: 155 Hz, Harmonics present?: Yes
APE4R4	Fundamental frequency average value: 155.87 Hz, Harmonics present?: Yes
AZWRHV	Fundamental frequency average value: 180, Harmonics present?: Yes
BU6ZTY	Fundamental frequency average value: 180, Harmonics present?: Yes
CVDERT	Fundamental frequency average value: 8.44 Hz, Harmonics present?: Yes
FVJZUP	Fundamental frequency average value: 155.29, Harmonics present?: Yes
GFQLDW	Fundamental frequency average value: 180 Hz, Harmonics present?: Yes
GNZAQV	Fundamental frequency average value: 179.86 Hz, Harmonics present?: Yes
HU2LMR	Fundamental frequency average value: approx 179hz, Harmonics present?: Yes
HX2XJR	Fundamental frequency average value: 180Hz as measured, possibly derived from 60Hz mains interference. multiples of 60Hz not detected. Harmonics only noted at 180Hz intervals., Harmonics present?: Yes
J3UD9Q	Fundamental frequency average value: approximately 180 Hz, Harmonics present?: Yes
LAN7UQ	Fundamental frequency average value: 180 Hz, Harmonics present?: Yes

TABLE 1

Question 1- 7 : Examination Questions	
WebCode	**Response** ** Inconsistencies not highlighted; No consensus achieved **
N79P6G	Fundamental frequency average value: 561.55, Harmonics present?: Yes
N7GLJM	Fundamental frequency average value: 264.26, Harmonics present?: No
PDQBXL	Fundamental frequency average value: 180 Hz, Harmonics present?: Yes
T94UVJ	Fundamental frequency average value: 180 Hz, Harmonics present?: Yes
UAUK9B	Fundamental frequency average value: 180 Hz, Harmonics present?: Yes
UP9QJB	Fundamental frequency average value: 180 Hz, Harmonics present?: Yes
X4F2LE	Fundamental frequency average value: 177 Hz, Harmonics present?: Yes
YPVLHE	Fundamental frequency average value: 180Hz, Harmonics present?: Yes
Z2HGN6	Fundamental frequency average value: 180 Hz, Harmonics present?: Yes
ZUY4TC	Fundamental frequency average value: 8.44 Hz, Harmonics present?: Yes

Question 1-7: What is the average value of the fundamental frequency of the dominant tonal noise present? Are harmonics of this tone present?

Consensus Result: A consensus was not achieved for this question.

TABLE 1

Question 1- 8 : Examination Questions

Question 1-8: Fundamental frequency average value:

Manufacturer's Response: 179.8 Hz / 180 Hz

WebCode	Response ** Inconsistencies not highlighted; No consensus achieved **
38J7PA	180 Hz
44DZM9	60 Hz
44HGK8	
73WPM8	Mean F0 = 170 Hz
7VUMEZ	179,215 Hz
937E24	179.8 Hz
9KQ6Z3	180
9LH8KX	180Hz
9LLM26	180 Hz
AD27MZ	155 Hz
APE4R4	155.87 Hz
AZWRHV	180
BU6ZTY	180
CVDERT	8.44 Hz
FVJZUP	155.29
GFQLDW	180 Hz
GNZAQV	179.86 Hz
HU2LMR	approx 179hz
HX2XJR	180Hz as measured, possibly derived from 60Hz mains interference. multiples of 60Hz not detected. Harmonics only noted at 180Hz intervals.
J3UD9Q	approximately 180 Hz
LAN7UQ	180 Hz
N79P6G	561.55
N7GLJM	264.26
PDQBXL	180 Hz
T94UVJ	180 Hz
UAUK9B	180 Hz
UP9QJB	180 Hz
X4F2LE	177 Hz

TABLE 1

Question 1- 8 : Examination Questions		
WebCode	**Response**	** Inconsistencies not highlighted; No consensus achieved **
YPVLHE	180Hz	
Z2HGN6	180 Hz	
ZUY4TC	8.44 Hz	

Question 1-8: Fundamental frequency average value:

Consensus Result: A consensus was not achieved for this question. Approximately 67% of responding participants reported the manufacturer's response.

TABLE 1

Question 1- 9 : Examination Questions

Question 1-9: Harmonics present?

Manufacturer's Response: Yes

WebCode	Response
38J7PA	Yes
44DZM9	Yes
44HGK8	Yes
73WPM8	Yes
7VUMEZ	Yes
937E24	Yes
9KQ6Z3	Yes
9LH8KX	Yes
9LLM26	Yes
AD27MZ	Yes
APE4R4	Yes
AZWRHV	Yes
BU6ZTY	Yes
CVDERT	Yes
FVJZUP	Yes
GFQLDW	Yes
GNZAQV	Yes
HU2LMR	Yes
HX2XJR	Yes
J3UD9Q	Yes
LAN7UQ	Yes
N79P6G	Yes
N7GLJM	No
PDQBXL	Yes
T94UVJ	Yes
UAUK9B	Yes
UP9QJB	Yes
X4F2LE	Yes

TABLE 1

Question 1- 9 : Examination Questions	
WebCode	**Response**
YPVLHE	Yes
Z2HGN6	Yes
ZUY4TC	Yes

Question 1-9: Harmonics present?

Consensus Result: Yes

Forensic Audio Enhancement Responses

TABLE 2

Question 2- 1 : Enhanced Audio Examination

Question 2-1: Describe the workflow used to clarify the audio, including specific software and filters applied.

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

WebCode	Response
38J7PA	The file 01 Track 1.wma was opened in Adobe Audition (build 24.6.0.69). All next steps were done in Audition. Conversion to the Audition internal working format (32-bit floating point PCM WAV) was automatically done during opening of the file. In addition to auditory analysis (listening on high quality loudspeakers and headphones), the Spectral Frequency Display (spectrogram analysis) and Frequency Analysis (spectrum analysis) tools were used during analysis and enhancement steps. First, the different audio channels were analyzed. The file 01 Track 1.wma has two audio channels (a stereo audio file, with a left and a right channel). The right channel does not contain any speech information, only low-level noise and audio artefacts. The left channel contains speech information and clearly audible noise. The audio file was split into two separate mono audio files, where the following enhancement steps were applied on the audio from the original left channel. A Parametric EQ with the following setting was applied to remove low and high frequency noise: High Pass 24 dB/Oct @ 140 Hz, Low Pass 36 dB/Oct @ 3500 Hz. A series of Notch Filters were used to suppress noise at specific frequencies. Clearly audible improvement was achieved with the following notch filter settings: 180 Hz -36 dB; 360 Hz -30 dB; 540 Hz -30 dB; 720 Hz -20 dB; 900 Hz -20 dB; 1080 Hz -20 dB. Additional audible improvement was achieved with the following notch filter settings: 139 Hz -20 dB; 168 Hz -20 dB; 191 Hz -20 dB; 221 Hz -20 dB; 348 Hz -15 dB; 371 Hz -15 dB. Additional improvement (less audible, but visible in spectrum and spectrogram) was achieved with the following notch filter settings: 1260 Hz -15 dB; 1439 Hz -15 dB; 1619 Hz -10 dB; 1796 Hz -10 dB; 2510 Hz -10 dB. Next, a Noise profile was created from a part of the audio file where no speech and only noise was audible: time interval for the noise profile: 4:11.255 till 4:12.989. With the Noise Reduction tool the noise in the recording was suppressed with the following settings: Noise Reduction: 40%, Reduce by: 6 dB. To suppress the peak at 0:00.45, a small fade-in, ending at 0:00.54, was applied. The amplitude of the audio was normalized to -0.5 dBFS. The audio was saved as 32-bit PCM WAV file with name "01 Track 1_L_enhanced_copy.wav". The original sampling frequency (44100 Hz) was not changed during the enhancement process and saving of the file. The SHA-256 value of the file 01 Track 1_L_enhanced_copy.wav is: 685aaff18dd1a9a2ee0291f41650938f65e56d25289427f9fcbeaf4021fa9e0c. All steps were documented. A second researcher listened to the original audio and the final enhanced audio and made some independent observations, which were discussed with the first researcher (four-eyes / four-ears principle).
44DZM9	Performed virus/malware scan. No threats found. Made working copy. All work performed on working copy. Unzipped file "24-5591%20Audio.zip" using 7zip. The .zip folder contained one (1) audio file named "01 Track 1.wma" with SHA256: 58A3049158D544B9D0D4661749B6A53DB2E010B6D9C57FA2BA03220F4B232F94 Performed aural review, critical listening, waveform and spectral analysis using iZotope RX11 Advanced. Right channel had no speech signal. Left channel suffered from severe hum and harmonics. Challenges affecting intelligibility lasted entire duration of recording. Opened file "01 Track 1.wma" in Adobe Audition 24.6.0.69. Split channels to mono and processed left channel only. Applied the following global filters: -iZotope RX11 De-hum -Waves Clarity Vx Pro Mono -Parametric Equalizer. Settings saved to screenshots below and as separate .PNG file (Note: SGWDE Best Practices for Enhancement of Digital Audio v1.2, Section 4.3.1.1 says that screenshots may serve as documentation of settings): See attached Audition_Settings.png. Compared input to output for all filters. Normalized to -1.0 dBFS. Saved as the following monaural WAV PCM file at 16-bit, 44.1 kHz: 01 Track 1_Enhanced.wav SHA256: 1B49F3422D0B6F5D24B6191754D875D23BF7282084B4C4F39B992688CFDBA525. Performed final review comparing result to unprocessed original.

TABLE 2

Question 2- 1 : Enhanced Audio Examination

WebCode	Response
44HGK8	1. Format converting from .wma to .wav using formatfactory. 2. Capture noise print and noise reduction filter using Audacity software. 3. Denoise filter using Adobe Audition software. 4. capture noise print and noise reduction filter using Adobe Audition software. 5. amplify using Adobe Audition software.
73WPM8	Audio Enhancement process: 1. Channel Separation: The right channel was removed, and the left channel was selected for enhancement as it contained clearer speech data. 2. Critical Listening: Conducted an auditory review of the audio to identify masking noise, clipping distortion, and volume inconsistencies. 3. Apply Filters: A. Broadband Noise Filter: Used to reduce masking noise, with a suppression depth of 40% and an adaptation time of 0.2 seconds. Whitening was enabled to retain natural tonal balance. B. Clip Restorer (applied twice): - First instance: Suppression depth set to 49%. - Second instance: Suppression depth set to 44%. - Corrected clipped sections to restore lost audio peaks. C. Amplifier: Adjusted the volume to ensure consistent loudness, set near 0 dB. 4. Output: - The enhanced audio was saved in uncompressed PCM Wave format. - The original sample rate and bit depth were maintained. - The output file was converted to Mono, preserving only the left channel containing speech. 5. Review and Documentation: - Conducted a final critical listening to confirm speech intelligibility. - Documented all steps, including filters and settings, for reproducibility and compliance with SWGDE Best Practices.
7VUMEZ	Software: iZotope RX 10 v10.2.0.1760 Tools: Spectral repair, Voice de-noise, Dialogue isloate
937E24	The audio was processed using iZotope RX Advanced v9.4.0 and the following filters: Mixing to duplicate the left channel to both channels of the audio, Normalize, EQ to reduce the hum, De-click, Voice De-noise, and Leveler. The processed audio was exported to a mono file.
9KQ6Z3	Software used: iZotope RX 10 Advanced Audio Editor v10.4.2.2124. Load file "01 Track 1.wma." There appears to be no intelligible audio data in the RIGHT channel. Highlight entirety of LEFT channel > Right click > Copy > File > New from Clipboard. Entire file time period: 00.00.00.000 – 00.10.07.062 / 0 – 26771456. Continue clarification on single channel. EQ: HP Brickwall 150 Hz Freq. LP 48 dB/oct 5545 Hz Freq. De-hum: Time Period to Learn. 00:05:11.000 – 00:05:12.765 / 13715100 – 13792935. Filter Parameters: Filter DC offset checked/Filter type: Static/Frequency: 180/Q: 1000/-60.0 dB/Harmonics: 14/Link harmonics: all/Slope: 0.50. Spectral De-noise: Time Period to Learn. 00:05:11.000 – 00:05:12.765 / 13715100 – 13792935. Filter Parameters: Threshold Noisy: -4.5/Threshold Tonal: -4.5/Reduction Noisy: 12.5/Reduction Tonal: 14.7/Quality: C/ Artifact control: 7.0/Smoothing: 2.0/Reduction curve checked/Algorithm Behavior Smoothing: 7.0 Algorithm: Extreme/Noise Floor Synthesis: 0.0/ Noise Floor Masking: 10.0/ Noise Floor Enhancement: 5.0/ Noise Floor Whitening: 10.0/Dynamics Knee: 1.5/Dynamics Release (ms): 80 De-wind: Reduction:4.0/Crossover frequency (Hz): 1200/Fundamental recovery: 6.0/Artifact smoothing: 3.5 Spectral De-noise (Adaptive): Adaptive mode checked/Learn time (s): 4.0/Threshold Noisy: -3.3/Threshold Tonal: -3.3/Reduction Noisy: 14.0/Reduction Tonal: 14.0/Quality: C/ Artifact control: 6.0/Smoothing: 0.0/Reduction curve checked (not active)/Algorithm Behavior Smoothing: 7.0 Algorithm: Extreme/Noise Floor Synthesis: 0.0/ Noise Floor Masking: 10.0/ Noise Floor Enhancement: 5.0/ Noise Floor Whitening: 10.0/Dynamics Knee: 1.5/Dynamics Release (ms): 80 Spectral De-noise: Time Period to Learn. 00:05:10.749 – 00:05:12.804/13704033 – 13794668. Filter Parameters: Threshold Noisy: -1.4/Threshold Tonal: -1.4/Reduction Noisy: 17.9/Reduction Tonal: 17.9/Quality: C/ Artifact control: 5.0/Smoothing: 0.0/Reduction curve checked (active)/Algorithm Behavior Smoothing: 7.0 Algorithm: Extreme/Noise Floor Synthesis: 0.0/ Noise Floor Masking: 10.0/ Noise Floor Enhancement: 5.0/ Noise Floor Whitening: 10.0/Dynamics Knee: 1.5/Dynamics Release (ms): 80 EQ: Band 1 Bell/425 Hz Freq/6.2 dB gain/0.5 Q. Leveler: Optimize for: Dialogue/Target level: -22.1/Responsiveness: 4.1/Preserve dynamics: 25. Normalize: Target peak level (dBFS): -1.00. File > Export. Wav/Bit depth: 16-bit/Dither: White noise (TPDF). For more information, please refer to the document "24-5591 Audio Clarification Documentation.pdf
9LH8KX	1. Converted the file to 44100 Hz 32-bit mono PCM .wav file. 2. Captured noise print and applied Noise reduction - Adobe Audition 2024 3. Normalized entire file to -0.1dB - Adobe Audition 2024 4. Applied Spectral De-noise to entire file - iZotope RX6 pluggin within Adobe Audition 2024 5. Applied a 'Hard Limiter' to -6dB - Adobe Audition 2024 6. Normalized entire file to -0.1dB - Adobe Audition 2024 I initially tried applying a notch filter, but the results were better with the options listed above.

TABLE 2

Question 2- 1 : Enhanced Audio Examination	
WebCode	**Response**
9LLM26	Adobe Audition: Extract the left channel from the original file. Apply filter to limit the voice signal band (300-3500 Hz) Apply Noise Reduction using the section 3:28 to 3:29 of the original file as the noise profile. Izotope RX: DE-HUM module to filter the effects of electrical noise. RUSTLE filter to eliminate crackles. LEVELER utility to level out signal variations optimized for Dialogue NORMALIZE to adjust the final signal level.
AD27MZ	Adobe Audition software used. Workflow - High Pass filter. Noise print captured from first few seconds then applied using noise reduction filter. Added some Gain. Extracted the left channel to a mono recording in WAV PCM format.
APE4R4	Software: Adobe audition 12.1.3.10; VoiceAI Cleaner™ Intelligent Audio Denoise Workstation 3.1.3 Workflow: 1.Analyze the voice to be inspected, and upon observing the spectrum, it is found that the voice content is concentrated in the left channel, with noise distributed across all frequency ranges. Simultaneously, there are stable harmonics at lower frequencies and an uneven distribution of harmonics at higher frequencies. Considering these issues, first, separate the left channel, then use inverse filtering to remove the stable harmonics at lower frequencies, followed by spectral subtraction to suppress other harmonics and background noise, and finally, enhance the required human voice through far-field noise reduction features. 2.Extract the audio file as a mono file and save the left channel as a separate recording file named "01 Track 1_L.wav". 3.Apply inverse filtering to the audio, set the filter length to 80 (range 1-100), and choose 10 iterations (range 1-50). 4.select the pure noise section from 04:45.245 to 04:46:237 seconds and perform noise capture. 5.Then carry out noise reduction on the entire audio based on spectral subtraction. 6.After that, enhance the required human voice through far-field noise reduction features. 7.Save the file.

TABLE 2

Question 2- 1 : Enhanced Audio Examination	
WebCode	**Response**
AZWRHV	1 x .zip file downloaded from CTS-portal: - 24-5591%20Audio.zip. Transferred from laptop to evidential PC via encrypted USB drive. SHA-1 hash taken of .zip file using 7-zip (v24.01). Hash matches hash provided by CTS-portal. Contents of .zip file extracted using 7-zip. 1 x .wma file contained inside: - 01 Track 1.wma. Audio quality assessment made using VLC (v3.0.16). L channel = High level static hum/buzz throughout duration of recording. High level broadband static noise throughout duration of recording. Low level bandwidth-limited male voice audible throughout. Sounds like voice recording is possibly a telephone or analogue radio recording. Listenability very low due to annoyance caused by low SNR. Speech partially intelligible. R channel = any audio is inaudible. Converted .wma file to 44.1kHz 16bit 2ch PCM .wav file using WinFF (FFMPEG 6.0 script) - extensive testing has validated this method of conversion within the lab. Our standard procedures require that we convert all audio to 24 bit however as this test has asked for the original bit depth to be maintained, I have converted this audio file to 16bit and subsequent processing will be completed at this bit depth. Audio quality and duration checks made. I believe the conversion has been satisfactory to go ahead with subsequent processing. Converted .wav opened in Cambridge Cedar (v14.0.2.490) and following processing applied: - Channel configuration: Discard right channel and use left channel to produce dual mono audio - DuBuzz: Reduce buzz/hum noise - DNS (Dialogue Noise Suppression): reduce background noise - NR5 (Trained to 9'30 - 9'40): reduce broadband noise - 2 x FNR: improve voice clarity - VoicEX: improve voice clarity - Adaptive Limiter: Boost level. Audio rendered through processing modules and processed .wav file exported out of Cedar as 44.1kHz 16bit 2ch PCM .wav. Processed .wav opened in iZotope RX (v10.3.0) and following processing applied: - EQ: High pass filter and low pass filter to reduce unwanted frequency content (speech is bandwidth limited) - Right hand channel discarded to give mono audio file (as requested) - Normalise: ensure audio is peaking at a suitable level (-0.5dBFS). See uploaded pdf doc for Cedar and RX processing settings. Overall I believe the processing to have made a large improvement to listenability. This version would greatly aid someone attempting to put together a transcript. This has mainly come from the reduction of the hum/buzz. There are however areas where the speech remains unintelligible. This is due to the poor SNR in the original recording as is unfortunately past the capabilities of the processing tools available. A peer review was completed by another competent practitioner. She had no suggestions for improvement. After her review we discussed the compromise between greater processing which would improve listenability but potentially at the expense of some intelligibility. Tests found that pushing the processing further caused audible artefacts in the speech and whilst being easier to listen to (particularly for long periods of time) there might have been some words that were more difficult to interpret. Therefore, for the purpose of this proficiency test I decided to be conservative with the processing. If this were a real case I would likely have made contact with the submitting party and provided them with an additional alternate version that had heavier processing so that they could use both versions to produce a transcript. Processed file exported out of iZotope RX: - 01 Track 1_Processed.wav. Output file inspected with MediaInfo (v22.03) - Duration: 10min 7s 62ms - 44.1kHz 16bit 1ch PCM .wav. 7-zip used to take SHA-256 hash of processed .wav file: ed8c7ebc97377abf7fc6f27bd18b73e8a62cc4484631c359f46efd36cd453e44 Processed .wav file and PDF containing processing settings copied to internet enabled laptop via encrypted USB drive for upload to CTS portal

TABLE 2

Question 2- 1 : Enhanced Audio Examination	
WebCode	**Response**
BU6ZTY	1. Received Forensic Proficiency testing 'Test 24-5591: Forensic Audio Analysis' from Supervisor. 2. Photographed provided Storage Device (SanDisk Ultra USB 3.0 32GB, Model SDCZ48-032G, SN: BM23120023944). Photographs available for view in case data. 3. Copied files from Storage Device to Forensic Data Case folder using: USB Write Blocker WB122, Analysis computer (Win10 x64) and TeraCopy version 3.17. a. USB Write Blocker WB122 was verified for use prior to connecting the Storage Device via a test USB file creation. b. Data copied from storage device was md5 hash generated and verified via TeraCopy 3.17. 4. Created a working copy of the original copied data. a. All files used from this point originate from working copy. b. All files md5 hash verified by TeraCopy 3.17. 5. Analyse audio file '01 Track 1.wma' using ffmpeg. a. //ffprobe -i "01 Track 1.wma" 6. Analyse audio file using iZotope RX9. Audio of relevance contained on left channel only. Copied left channel to new iZotope session. Disregarded right channel. 7. Enhance audio file '01 Track 1_left.wav' using iZotope RX9. Refer iZotope report for all filters and settings applied. Summary of clarification process: a. Document File information and Waveform Statistics. b. Remove identified DC offset using iZotope De-Hum. c. Remove hum of fundamental frequency (180hz) and harmonics using iZotope De-Hum. d. Fixed discontinuous waveforms using iZotope De-click. e. Low and high frequency cut using iZotope EQ.
CVDERT	Sound Cleaner II being used to clean an audio file using multiple filters. These include the Broadband Noise Filter to reduce background noise, Click Suppressor to eliminate sharp noises, Dynamic Range Control to balance volume, Equalizer to adjust frequencies, Tone Suppressor to reduce low-frequency hums, and Reverb Suppressor to minimize echoes.
FVJZUP	1. Open Adobe Audition. 2. Drag and drop audio file into the Files panel. 3. Double-click the audio file to open it in the Waveform Editor. 4. Play the audio and locate a portion where only noise is present. 5. Highlight this section. 6. Go to Effects > Noise Reduction/Restoration > Capture Noise 7. Click OK. 8. Highlight the entire audio file. 9. Go to Effects > Noise Reduction/Restoration > Noise Reduction (Process). 10. In the Noise Reduction panel: a. Adjust this slider to determine how much noise is removed. b. Set how much noise reduction is applied in decibels. 11. Repeat the same process for other noise that contain in the audio file. 12. After specific noise are removed, then go to Effects > Amplitude and Compression > Amplify to boost the speech in audio file. 13. Do the Normalization Process to remove the loudest peak to a specific level (I set it to -10 dB). 14. Save / Export the clean audio into WAV format.
GFQLDW	Process in adobe audition: 1. The channels were separated and the right channel was removed, as it did not contain any voice data. 2. Intense click reduction using the DeClicker filter. 3. Normal clip reduction using the DeClipper filter. 4. Scientific filter for subsonic vibration elimination (reduction of frequencies from 0 to 100 Hz) and reduction of frequencies from 8000 Hz to 22050 Hz to eliminate artifacts and other noises. 5. DeHummer filter for 180 Hz frequency and 8 harmonics at -35dB. 6. Band-reject filter at frequencies 140 Hz (-20dB), 220 Hz (-20dB), 1620 Hz (-15dB), 1800 Hz (-15dB), 2160 Hz (-10dB), and 2520 Hz (-10dB). 7. Noise reduction by 12% and 32dB, FFT size 2048. 8. Hiss reduction by 8dB, noise base 8dB, FFT size 2048. 9. Vocal amplifier filter. 10. Reverb cancellation at 40%. 11. Normalize to -1dB. The sample is saved in .wav format (PCM), in mono, with the original file's sampling rate and bit depth.
GNZAQV	We used the Cedar Cambridge Forensic System. The workflow is the following: CEDAR Cambridge report 24 Oct 2024, 09:30:30 Reference : Process manager chain: I/O sample rate : 44100.00; I/O channels : 1 PROCESS MANAGER Process Chain : channels : 1 sample rate : 44100.00 Debuzz-3: Current Settings : on/off: on; frequency : 179.86 Hz; threshold : 0.00 dB; reduction : -50.00 dB; bandwidth : 3000.00 Hz; detection channels : 1; tracking : on; harmonic mode : all. NR5: Current Settings : on/off: on; level : 0.00 dB; noise reduction; gain : -5.00 dB LS1: frequency : 100.00 Hz; gain : -50.00 dB; slope : 120.00 dB/octave. HS2: frequency : 4500.00 Hz; gain : -50.00 dB; slope : 120.00 dB/octave; noise free EQ; gain : 0.00 dB. LS1: frequency : 100.00 Hz; gain : -50.00 dB; slope : 120.00 dB/octave. HS2: frequency : 4500.00 Hz; gain : -50.00 dB; slope : 120.00 dB/octave. Trinity Enhance: Current Settings : on/off: on; motor : 0.00; noise : 11.17; background : 0.00; speech : 29.33. FNR: Current Settings : on/off: on; speed : 0.20 Hz; resolution : optimal; bias : 0.00 dB; attenuation : -4.00 dB; focus : 70.00. Dehiss-3: Current Settings : on/off: on; threshold : 80.00 dB; attenuation : -15.00 dB; brightness : 0.00. Adaptive Limiter 2: Current Settings : on/off: on; input gain : 6.00 dB; threshold : -1.00 dB; spectral : 0.00; HF temporal : 0.00; LF temporal : 0.00; oversample : off. MD5 hash value 01 Track 1_cleaned.wav: c4148f7b0ab5981e3918b6cf87a6be93

TABLE 2

Question 2- 1 : Enhanced Audio Examination	
WebCode	**Response**
HU2LMR	Downloaded ZIP file and confirmed SHA1 value matched expected above. File 01 Track 1.wma imported directly into iZotope RX 9. Appears to be 2ch file. However, no audio is present on Rch. File exported as 16bit WAV and converted to dual mono file. Resultant file reloaded in iZotope for processing. Recording contains speech with persistent hum. Occasional distortion present. Voice is masked by hum and very difficult to hear. Initial EQ applied 200Hz-4kHz to remove low end rumble and high end hiss. Hum not affected. Various notch and buzz filters attempted. Appears fundamental frequency is approx 179Hz and may drift slightly. iZotope D-Hum tool applied in dynamic adaptive mode which appears to attack hum adequately. Result is easier to listen to with hum level decreased. Dialogue Isolate filter applied which appears to make voice slightly more intelligible. Voice becomes very difficult to hear from approx 9min onwards and is effectively masked. Overall, resultant file is subjectively easier to listen to and more intelligible. However, voice is still masked at time. Resultant file converted to MONO file (as per CTS instructions), uploaded to CTS portal and attached to exam record.
HX2XJR	Zip folder "24-5591%20Audio" transferred from specified location. Unzip and extract the "01 Track 1.wma" file. Transfer the wma file to a workstation in the Audio laboratory. Open the file within Cedar Cambridge software, conduct aural examination, and apply filtering solution. Note that signal is only apparent in the left channel of the audio file. Mono processing applied. Filtering modules applied: Gain - to reduce level going into the processing chain. DeBuzz - to remove buzz apparent in the audio. Trinity Enhance - to reduce some of the broadband noise and increase speech level. EQ - to improve intelligibility of target speech. Compressor - to maintain a more consistent output level. Gain applied to output of compressor, to increase the overall output level. FNR - to further reduce broadband noise. Adaptive limiter applied to final output. Render and save resultant audio file: "01 Track 1_processed.wav"
J3UD9Q	The audio was imported into Adobe Audition and reviewed. The audio was in stereo, but the right channel did not contain speech. Extracted to mono and used the left channel for processing. Filters (including iZotope plugins) were applied to reduce noise and improve the intelligibility of the speech. Filters included: Parametric Equalizer, iZotope RX 10 De-hum, iZotope RX 10 De-crackle, iZotope RX 10 Spectral De-noise, iZotope RX 10 Voice De-noise, and Hard Limiter. The end of the file (approximately 00:08:49-00:10:07) had lower talker levels and higher noise than the rest of the file. Additional filters were applied to this portion, including: iZotope RX 10 Voice De-noise and Hard Limiter. The processed file was exported as a .wav file.
LAN7UQ	First, SHA256 was extracted from the audio. Then the audio was converted to mono WAVE file (i.e. the channel, which contains speech, was preserved) with FFmpeg. Then the converted audio was cleaned with iZotope RX 10 using following filters: Gain, Spectral repair, Spectral Denoise and Loudness Control. Finally, edited audio was saved as 16 bit 44.1 kHz WAVE file. New SHA256 was extracted from the edited audio.
N79P6G	I have used Sound Cleaner II software for enhancement and clarifying the given audio file. Firstly, i upload the original audio file in to the Sound Cleaner II software then go to edit tab the choose the Divide stereo to two mono option. This will result in display two channel for current audio file (Right and left channel). The left channel will contain the audio file and the right channel shows empty. After extraction the audio file to mono will be in one channel as its requested in the question above. For the reduce noise and enhancement I do the following. Firstly, choose the left channel and choose Broadband Noise filter from filters option in the FILTRATION SCHEME. Then turn ON the Suppression Depth and put it to Max (40 dB). Then I choose Reverb Suppressor from filters option in the FILTRATION SCHEME. Then turn ON the Suppression Depth and put it to Max (40 dB). Last I choose Amplifier from filters option in the FILTRATION SCHEME. Then turn ON the Suppression Depth and put it to Max (12 dB). Then save the changes to the audio file and save it as PCM Wave format.
N7GLJM	Load provided file into Adobe Audition. Review media information. There were found to be two channels. As the second channel did not contain any audio, the file was converted to mono. Applied the following filters in the following order: "Dehummer" (60hz), "Notch Filter", "Speech volume leveler." The enhanced file was written out as (1) WAV PC (44.1 khz and 32 bit).

TABLE 2

Question 2- 1 : Enhanced Audio Examination	
WebCode	**Response**
PDQBXL	I used Adobe(R) Audition v25.0.0.47 software to convert the WMA file to mono and apply an Effects Rack with the following filters: 1) adaptive iZotope RX 4 Denoiser. 2) BandPass filter of 100 Hz to 3500 Hz. 3) Notch Filter of -20 dB for 180 Hz, 360 Hz, 540 Hz, 720 Hz, 900 Hz, & 1080 Hz. 4) 2nd application of above Notch Filter. 5) Hard Limiter, with boost of 15 dB & maximum amplitude of -3 dB. I then applied a manual iZotope RX 4 Denoiser filter, with the Learned interval of 5:10.882 to 5:12.000. After applying the above filters, I exported the audio as a mono, 44,100 Hz, 16-bit Wave PCM file.
T94UVJ	1. 24-5591 Audio.zip file's hash verification. 2. Assessment: two channels, only one with speech signal; distortions: a) stationary noise with fundamental frequency on 180 Hz with harmonics b) broadband noise 3. Preprocessing: using ffmpeg (version N-117899-g3c3bf6c109-20241125) WMA was converted to PCM maintaining original sampling rate and bit depth, next channels were splited; for further processing only left channel was used to enhanced due to lack of speech in right channel. 4. Audio enhancement: Applied software: Adobe Audition CS6 v.5 (Noch filter, FFT filter); Izotope RX v.10, (Dialogue Isolate, Spectral De-noise, Normalize). 5. Hash SHA-256 of the file "01 Track 1_mono_U2726A_T94UVJ_final_version.wav": 2A3052834E45146A6B87773B7122E6CDC8C38371088A1A95F0F2ED4954874C76.
UAUK9B	Software Used: Audacity version 3.7.0. Tasks listed in Part 2 were performed and one enhanced audio file named "enhanced audio.wav" was exported and is being provided on the accompanying DVD. 2-1 Workflow used to clarify the audio, including specific software and filters for the audio enhancement is as follows: • Audio was enhanced using Audacity version 3.7.0; • The file was imported in Audacity. All the audio was selected, and Notch filter was applied. Notch filter Parameters: Frequency (Hz) : 180,359,540,719,899,1079,1259,1439 and 1618 Q : 40 • A section of recording containing just noise was selected and the noise profile was generated using Noise Reduction filter. After that all the audio was selected and Noise was reduced. Noise Reduction Parameters: Noise reduction (db): 15; Sensitivity: 6.0; Frequency smoothing (bands): 6. • Graphic EQ filter was applied. Graphic EQ Parameters: A boost of 6db was applied to these frequencies (315,400,500,630,800 and 1K) Hz
UP9QJB	The .wma file was processed in "iZotope RX 10 Advanced (64-bit) v10.3.0.1775", "Sound Cleaner II" and "VoiceClean Intelligent Audio Denoise Workstation". iZotope RX 10 Advanced: -opened the file, selected the left-channel and copied, then created mono file with 'New from Clipboard'. -'De-hum' module to reduce tonal noise. Adaptive mode (selected), Filter type (Dynamic), Sensitivity 3.0, Bands 20, Filter Q 340. -'Dialogue Isolate' module to attenuate background noise. Dialogue gain 0 dB, Noise gain -12 dB, Sensitivity 8.0, Ambience preservation 50, Quality (Best). Sound Cleaner II: -'Equalizer' module to filter out unwanted high-frequency information. Spectrum (Accumulate), Sclae (Log), FFT (2048), Inverse Filter, Threshold 3000 Hz. VoiceClean Intelligent Audio Denoise Workstation: -'Broadband Denoising' module to reduce background noise. Suppresion Strength 18 dB. The rendered audio was exported from VoiceClean as a 16 bit, 44.1 kHz, mono channel PCM wav file.
X4F2LE	Using Adobe Audition v.24.6.0.69, I determined the tone to be harmonic. Below are the steps taken for enhancement: 1) Notch filter 177 Hz increments 2) De-click: highlight click, lower amplitude to -16.2 dB 3) De-hum 177 Hz increments, 8 harmonics 4) De-noise 60% all frequencies 5) De-reverb 10% low frequencies 6) Amplify +10 dB 7) Normalize 0.3 dB Saved derivative as an uncompressed PCM Wave format, maintaining original file sample rate of 44.1 kHz and bit depth of 16 bits, and one channel mono, containing speech data.
YPVLHE	Using Adobe Audition 2024: Left channel was exported as a mono audio track for enhancement. 32-Bit was chosen because the 16-bit option was not available. Frequency make up of the voice lies primarily between 200Hz – 2000Hz. Using iZotope RX10 Advanced: The De-Hum filter was used starting at 180Hz and covering 13 harmonics to reduce the humming noise. Using the EQ filter, a high pass filter was placed at 200Hz, and a low pass filter was placed at 2117Hz. The Spectral De-Noise filter was used to reduced broadband noise in the recording. The EQ filter was used again to placed notches at 220Hz and 660Hz starting at 8:28 in the recording to reduce the buzzing noise. Clip gain was increased by +5.5dB for the whole recording. Exported as a 16-bit, 44.1kHz uncompressed .WAV audio file.

TABLE 2

Question 2- 1 : Enhanced Audio Examination	
WebCode	**Response**
Z2HGN6	To reduce the fundamental frequency of the dominant tonal noise the Brush selection tool (v 6) in Izotope RX 9 (v 5) was used. This was done in ten steps starting with the lowest frequency, to then continuing trying to remove the following nine found harmonics. After this the EQ (v 7) was used to apply a low pass filter set to 4500 Hz. This was done since little to none of the speech frequency could be found above this value. Further on the Spectral De-noise (Adaptive mode) (v 8) was used to reduce additional masking frequencies of the dominant tonal noise. To enhance the clarity of the speech an additional EQ (v 7) was applied. This time a comb filter was used to increase the speech frequencies that could be found in between the dominating tonal noise and its harmonics. After this the audio file was opened in Wavelab Pro (v 2) processed with the Q8 (v 3). This to enhance the significant middle-frequency for speech and thereby increase the speech intelligibility furthermore. Finally, the L1 Limiter (v 4) was applied to even out the dynamic range of the overall audio and thereby one last time increase the audibility of speech. The resulting file, 01_Track 1_Enhanced.wav, was saved as a Wave(PCM), 16 bit, fs. 44,1 kHz, mono file.
ZUY4TC	The Reference Noise (RN) Canceller removed interference from external signals, while an Equalizer adjusted specific frequency bands for better tonal balance. A Broadband Noise Filter eliminated consistent background noise across all frequencies, and Dynamic Range Control ensured consistent volume by managing peaks and amplifying quieter sections. The Reverb Suppressor minimized echo and reverberation, and finally, an Amplifier boosted the overall audio signal, resulting in a cleaner, more refined output.

Question 2-1: Describe the workflow used to clarify the audio, including specific software and filters applied.

Consensus Result: This was a free form question on methods and tools used. No consensus response is provided.

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
38J7PA	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). The participant saved the file in PCM Wave format, maintaining the original 44.1kHz sample rate, but bit depth was increased to 32-bit float.
44DZM9	Submitted enhanced file(s) were reviewed by an expert who confirmed that this participant completed all requested audio enhancement steps.
44HGK8	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). The participant saved the file as 2-channel audio with the essentially blank Right Channel included.
73WPM8	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). The participant saved the file in PCM Wave format, maintaining the original 44.1kHz sample rate, but bit depth was increased to 32-bit float.
7VUMEZ	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). File was saved as 2-channel audio with the essentially blank Right Channel included. While noise was significantly reduced, the speech sounds more "muffled" due to over-processing, with filtering artifacts being prominent.
937E24	Submitted enhanced file(s) were reviewed by an expert who confirmed that this participant completed all requested audio enhancement steps.
9KQ6Z3	Submitted enhanced file(s) were reviewed by an expert who confirmed that this participant completed all requested audio enhancement steps.
9LH8KX	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). The participant saved the file in PCM Wave format, maintaining the original 44.1kHz sample rate, but bit depth was increased to 32-bit float.
9LLM26	Submitted enhanced file(s) were reviewed by an expert who confirmed that this participant completed all requested audio enhancement steps.
AD27MZ	Submitted enhanced file(s) were reviewed by an expert who confirmed that this participant completed all requested audio enhancement steps with the following note(s). While overall noise was reduced, gain was increased too much resulting in clipping artifacts.
APE4R4	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). The participant saved the file in PCM Wave format, maintaining the original 44.1kHz sample rate, but bit depth was increased to 32-bit float.

TABLE 3

Forensic Audio Enhancement Observations	
WebCode	**Observational Notes**
AZWRHV	Submitted enhanced file(s) were reviewed by an expert who confirmed that this participant completed all requested audio enhancement steps.
BU6ZTY	Submitted enhanced file(s) were reviewed by an expert who confirmed that this participant completed all requested audio enhancement steps.
CVDERT	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). While the participant saved the file with a ".wav" extension, it is a MPEG-4/AAC LC format file instead of the PCM Wave that was requested.
FVJZUP	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). File was saved as 2-channel audio with the essentially blank Right Channel included. While overall noise was reduced, gain was increased too much resulting in clipping artifacts.
GFQLDW	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). The participant saved the file in PCM Wave format, maintaining the original 44.1kHz sample rate, but bit depth was increased to 32-bit float.
GNZAQV	Submitted enhanced file(s) were reviewed by an expert who confirmed that this participant completed all requested audio enhancement steps.
HU2LMR	Submitted enhanced file(s) were reviewed by an expert who confirmed that this participant completed all requested audio enhancement steps.
HX2XJR	Submitted enhanced file(s) were reviewed by an expert who confirmed that this participant completed all requested audio enhancement steps.
J3UD9Q	Submitted enhanced file(s) were reviewed by an expert who confirmed that this participant completed all requested audio enhancement steps.
LAN7UQ	Submitted enhanced file(s) were reviewed by an expert who confirmed that this participant completed all requested audio enhancement steps.
N79P6G	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). The participant saved the file in PCM Wave format, maintaining the original 44.1kHz sample rate, but bit depth was increased to 32-bit float. While overall noise was reduced, gain was increased too much resulting in clipping artifacts.
N7GLJM	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). The participant saved the file in PCM Wave format, maintaining the original 44.1kHz sample rate, but bit depth was increased to 32-bit float. The fundamental frequency of the tonal noise in this recording is approx. 180Hz. By using the default Base Filter Frequency setting of 60Hz in DeHummer (likely with 8 harmonics selected), noise bands were attenuated from 60Hz to 480Hz in 60Hz increments. This attacked the first two noise bands at 180Hz and 360Hz but left all the other 180Hz harmonics above 480Hz. Setting DeHummer to attack the fundamental frequency of 180Hz would have attenuated harmonics up to 1440Hz. Additional noise reduction filters could be used to increase intelligibility. The Notch filter applied at approx. 5kHz had no effect on noise or intelligibility.
PDQBXL	Submitted enhanced file(s) were reviewed by an expert who confirmed that this participant completed all requested audio enhancement steps.
T94UVJ	Submitted enhanced file(s) were reviewed by an expert who confirmed that this participant completed all requested audio enhancement steps with the following note(s). While noise was significantly reduced, the speech sounds more "muffled" due to over-processing, with filtering artifacts being prominent.
UAUK9B	Submitted enhanced file(s) were reviewed by an expert who confirmed that this participant completed all requested audio enhancement steps.

TABLE 3

Forensic Audio Enhancement Observations	
WebCode	**Observational Notes**
UP9QJB	Submitted enhanced file(s) were reviewed by an expert who confirmed that this participant completed all requested audio enhancement steps.
X4F2LE	Submitted enhanced file(s) were reviewed by an expert who confirmed that this participant completed all requested audio enhancement steps.
YPVLHE	Submitted enhanced file(s) were reviewed by an expert who confirmed that this participant completed all requested audio enhancement steps.
Z2HGN6	Submitted enhanced file(s) were reviewed by an expert who confirmed that this participant completed all requested audio enhancement steps.
ZUY4TC	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). The participant saved the file in PCM Wave format, maintaining the original 44.1kHz sample rate, but bit depth was increased to 32-bit float.

Additional Comments

TABLE 4

WebCode	Additional Comments
44DZM9	For question 1-7, the fundamental frequency had low amplitude and was difficult to measure. The harmonics were much stronger. However, based on even and odd-ordered harmonic multiples, the fundamental average is 60 Hz. The poor frequency response of the microcassette likely rolled off most of the lower frequency content.
9LH8KX	I have also taken screenshots of all the filters used and the settings applied if required
AD27MZ	I applied the repair assistant in Izotope and it also did a good job. Left a bit of noise behind but very useful.
APE4R4	Hope the testing items can be released before July.
AZWRHV	Unable to check hash of uploaded file but would recommend hash is checked on download of processed .wav file. On delivery of processed file would advise receiving party attends the audio lab where they will be able to use a good quality pair of headphones in a quiet listening environment to listen to audio.
HU2LMR	Jacksum Name: 01 Track 1.wma Size: 9793895 bytes (9564 KiB) SHA256: 58A3049158D544B9D0D4661749B6A53DB2E010B6D9C57FA2BA03220F4B232F94 FFMPEG v7.1 FFPROBE Input #0, asf, from '.\01 Track 1.wma': Metadata: title : Track 1 album : Unknown album (8/22/2016 10:51:04 AM) WM/Track : 0 track : 1 WMFSDKVersion : 12.0.22621.3527 WMFSDKNeeded : 0.0.0.0000 IsVBR : 0 DeviceConformanceTemplate: L1 Duration: 00:10:07.06, start: 0.000000, bitrate: 129 kb/s Stream #0:0(eng): Audio: wmav2 (a[1][0][0] / 0x0161), 44100 Hz, 2 channels, fltp, 128 kb/s MediaInfo v24.06 General Complete name : 01 Track 1.wma Format : Windows Media File size : 9.34 MiB Duration : 10 min 7 s Overall bit rate mode : Constant Overall bit rate : 129 kb/s Maximum Overall bit rate : 129 kb/s Album : Unknown album (8/22/2016 10:51:04 AM) Track name : Track 1 Track name/Position : 1 Encoded date : 2024-05-01 23:06:24 UTC Audio ID : 1 Format : WMA Format version : Version 2 Codec ID : 161 Codec ID/Info : Windows Media Audio Description of the codec : Windows Media Audio 9.2 - 128 kbps, 44 kHz, stereo 1-pass CBR Duration : 10 min 7 s Bit rate mode : Constant Bit rate : 128 kb/s Channel(s) : 2 channels Sampling rate : 44.1 kHz Bit depth : 16 bits Stream size : 9.26 MiB (99%) Language : English (United States)
N79P6G	I don't have any additional comment on this. I have submit all answers and description.
YPVLHE	A second pass of the spectral de-noise was not used to reduced the broadband noise surrounding the voice because it started to eliminate the sibilance in the words. EQ, voice/dialogue isolate filters were also affecting the voice. I did not want to over process the recording to the point of affecting what was being said in the recording.
Z2HGN6	Exact values and settings on each plug in used can be sent upon request.

-End of Report-
(Appendix may follow)

Collaborative Testing Services ~ Forensic Testing Program

Test No. 24-5591: Forensic Audio Analysis

DATA MUST BE SUBMITTED BY **Dec. 02, 2024, 11:59 p.m. EST** TO BE INCLUDED IN THE REPORT

Participant Code: U1234A WebCode: YK3GPB

Test Instructions:

1. Apply filtration to reduce masking noise and increase the intelligibility of speech in the recording.
2. Guidelines described in section 4 of the SWGDE Best Practices for the Enhancement of Digital Audio should be followed.
3. Answer questions regarding the audio file properties, attributes of the audio content, and procedures to clarify the speech.

Scenario:

A microcassette recording was made of an attorney calling in to a disciplinary hearing. Later, the microcassette was digitized into a .wma file (“01 Track 1.wma”). The submitter of this recording stated that the original microcassette is no longer available, but they need the speech in the digitized copy clarified in order to have it accurately transcribed.

This test is designed to measure your knowledge and skill in the following digital forensic audio processes: Data verification, Media characterization, Audio processing, and Audio enhancement. These are based on the competencies outlined in section 3.6 of the SWGDE Core Competencies for Forensic Audio.

Evidence:

To verify a complete and accurate download, use the tool of your choice to verify the integrity of the file.
24-5591 Audio.zip MD5 hash value: 197a7ddbd46ab1a852913f2a79126bf6
24-5591 Audio.zip SHA1 hash value: 68ab78c0c1a33fb59313fa216767a4679a50301f

Participant Code: U1234A
WebCode: YK3GPB

1-1). What is the SHA-256 hash value of the 01 Track 1.wma file?

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

1-3). How many audio channels are contained in the audio file? Report a numeric value.

1-4). What is the length in time of the audio file?

1-5). What is the bit depth of the audio file?

1-6). What is the sample rate of the audio file?

1-7). What is the average value of the fundamental frequency of the dominant tonal noise present? Are harmonics of this tone present?

Fundamental frequency average value:

Harmonics present?

Yes ○ No ○

Part 2: Enhanced Audio Examination

A. Process the audio using filters to reduce noise and increase the intelligibility of speech.
B. Save the clarified audio in the uncompressed PCM Wave format, maintaining the original file sample rate and bit depth. It is expected that only the channel containing speech data will be saved in the enhanced copy (The resulting file should be one channel/Mono).

Uploaded file name:

2-1). Describe the workflow used to clarify the audio, including specific software and filters applied.

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 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 ANAB and/or A2LA. (Accreditation Release section below must be completed.)
○ This participant's data is **not** intended for submission to 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.

A2LA Certificate No.

Step 2: Complete the Laboratory Identifying Information in its entirety

Authorized Contact Person and Title

Laboratory Name

Location (City/State)