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FORENSIC TESTING PROGRAM

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Forensic Audio Analysis
Test No. 25-5591 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 39 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

This 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: The digital audio was acquired via a predetermined, staged event involving a recording of a voicemail and compiled into a .m4a zipped file. Following sample validation, the file was uploaded to the CTS Portal for download by test participants. A MD5 and SHA1 digests (cryptographic checksum, or 'hash') were calculated for the compressed data and provided to participants to enable validation of a 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 structured the final questions utilized in this test. The following tools were utilized in the validation of this test: QuickHash-GUI (version 3.3.4), MediaInfo (version 25.04), iZotope RX 11 Advanced Audio Editor, and Audacity (version 3.7.4). CTS does not endorse any particular tools.

SCENARIO PROVIDED TO PARTICIPANTS

A person of interest (POI) claims that he was home alone during a specific time window relevant to an ongoing investigation. However, a voicemail stored on a third party's phone reveals that the POI's cellular phone, during that same period, inadvertently placed a call from a pocket during a one-on-one basketball game at a public court.

Investigators would like to examine the voicemail recording to determine whether any speech or background sounds captured during the call contradict the individual's stated alibi. Audio clarification processing is needed to improve listenability and increase intelligibility through noise reduction and speech clarification.

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Manufacturer's Information, continued

Question — Manufacturer's Response - Examination Questions

1-1 What is the SHA-256 hash value of the voicemail-2095.m4a file?

Manufacturer's Response:

97495995C89FB18006DF8104F1F44DAFC162A1163BD91E4C53CF03B6575B74EB

1-2 What is the audio format of this MPEG-4 container file?

Manufacturer's Response:

Advanced Audio Coding (or Codec) Low Complexity or AAC LC or AAC

1-3 How many audio channels are contained in the audio file? Provide a NUMERIC response.

Manufacturer's Response:

1

1-4 What is the length in time of the audio file? Provide your answer in the following format: MM:SS.

Manufacturer's Response:

03:00

1-5 What is the sample rate of the audio file? Provide a NUMERIC response and the associated units.

Manufacturer's Response:

8 kHz (8000 Hz)

1-6 What is the frequency value of the tonal noise present? Provide a NUMERIC response and the associated units.

Manufacturer's Response:

Frequency value/Units: 1.2 kHz (1200 Hz)
Are harmonics of this tone present? No

1-7 Are any clipped audio samples present in the file?

Manufacturer's Response:

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.

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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. Refer to Manufacturer's Information for preparation details.

A total of 39 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 seven examination questions achieved consensus. 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.

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Forensic Audio Examination Responses

TABLE 1

Question 1- 1 : Examination Questions

Question 1-1: What is the SHA-256 hash value of the voicemail-2095.m4a file?

Manufacturer's Response: 97495995C89FB18006DF8104F1F44DAFC162A1163BD91E4C53CF03B6575B74EB

WebCode	Response
4BPU69	97495995c89fb18006df8104f1f44dafc162a1163bd91e4c53cf03b6575b74eb
4KV2K9	97495995c89fb18006df8104f1f44dafc162a1163bd91e4c53cf03b6575b74eb
4VEDX8	97495995c89fb18006df8104f1f44dafc162a1163bd91e4c53cf03b6575b74eb
6KCCX8	97495995c89fb18006df8104f1f44dafc162a1163bd91e4c53cf03b6575b74eb
7B2AC8	97495995c89fb18006df8104f1f44dafc162a1163bd91e4c53cf03b6575b74eb
7YALG8	97495995C89FB18006DF8104F1F44DAFC162A1163BD91E4C53CF03B6575B74EB
8QCBL3	97495995c89fb18006df8104f1f44dafc162a1163bd91e4c53cf03b6575b74eb
9PWEU3	97495995C89FB18006DF8104F1F44DAFC162A1163BD91E4C53CF03B6575B74EB
CK6D4X	97495995C89FB18006DF8104F1F44DAFC162A1163BD91E4C53CF03B6575B74EB
CN9ANZ	97495995c89fb18006df8104f1f44dafc162a1163bd91e4c53cf03b6575b74eb
CUU2YZ	97495995c89fb18006df8104f1f44dafc162a1163bd91e4c53cf03b6575b74eb
EXM6GW	97495995c89fb18006df8104f1f44dafc162a1163bd91e4c53cf03b6575b74eb
F6YLNW	97495995c89fb18006df8104f1f44dafc162a1163bd91e4c53cf03b6575b74eb
GW6VGV	97495995c89fb18006df8104f1f44dafc162a1163bd91e4c53cf03b6575b74eb
JBVGZR	97495995C89FB18006DF8104F1F44DAFC162A1163BD91E4C53CF03B6575B74EB
JEUTVR	97495995C89FB18006DF8104F1F44DAFC162A1163BD91E4C53CF03B6575B74EB
JKL37Q	97495995C89FB18006DF8104F1F44DAFC162A1163BD91E4C53CF03B6575B74EB
JRHZTT	97495995C89FB18006DF8104F1F44DAFC162A1163BD91E4C53CF03B6575B74EB
JWEJVR	97495995C89FB18006DF8104F1F44DAFC162A1163BD91E4C53CF03B6575B74EB
JXA3WT	97495995c89fb18006df8104f1f44dafc162a1163bd91e4c53cf03b6575b74eb
KJNALR	97495995c89fb18006df8104f1f44dafc162a1163bd91e4c53cf03b6575b74eb
LNANPN	97495995c89fb18006df8104f1f44dafc162a1163bd91e4c53cf03b6575b74eb
M84GBM	97495995c89fb18006df8104f1f44dafc162a1163bd91e4c53cf03b6575b74eb
M9ELLP	97495995c89fb18006df8104f1f44dafc162a1163bd91e4c53cf03b6575b74eb
MFYM4N	97495995c89fb18006df8104f1f44dafc162a1163bd91e4c53cf03b6575b74eb
MND7MM	97495995C89FB18006DF8104F1F44DAFC162A1163BD91E4C53CF03B6575B74EB
MQUVMQ	97495995c89fb18006df8104f1f44dafc162a1163bd91e4c53cf03b6575b74eb

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TABLE 1

Question 1- 1 : Examination Questions

WebCode	Response
PD7AHM	97495995c89fb18006df8104f1f44dafc162a1163bd91e4c53cf03b6575b74eb
PK6UJH	97495995C89FB18006DF8104F1F44DAFC162A1163BD91E4C53CF03B6575B74EB
Q29QFM	97495995c89fb18006df8104f1f44dafc162a1163bd91e4c53cf03b6575b74eb
Q2LH7H	97495995C89FB18006DF8104F1F44DAFC162A1163BD91E4C53CF03B6575B74EB
R9ZQ8K	97495995C89FB18006DF8104F1F44DAFC162A1163BD91E4C53CF03B6575B74EB
T9EARH	97495995C89FB18006DF8104F1F44DAFC162A1163BD91E4C53CF03B6575B74EB
U7PMAE	97495995c89fb18006df8104f1f44dafc162a1163bd91e4c53cf03b6575b74eb
UEZPVE	97495995c89fb18006df8104f1f44dafc162a1163bd91e4c53cf03b6575b74eb
VLFPQH	97495995c89fb18006df8104f1f44dafc162a1163bd91e4c53cf03b6575b74eb
YW8DBB	97495995c89fb18006df8104f1f44dafc162a1163bd91e4c53cf03b6575b74eb
YZ389C	97495995c89fb18006df8104f1f44dafc162a1163bd91e4c53cf03b6575b74eb
Z2CVL9	97495995C89FB18006DF8104F1F44DAFC162A1163BD91E4C53CF03B6575B74EB

Question 1-1: What is the SHA-256 hash value of the voicemail-2095.m4a file?

Consensus Result: 97495995C89FB18006DF8104F1F44DAFC162A1163BD91E4C53CF03B6575B74EB

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TABLE 1

Question 1- 2 : Examination Questions

Question 1-2: What is the audio format of this MPEG-4 container file?

Manufacturer's Response: Advanced Audio Coding (or Codec) Low Complexity or AAC LC or AAC

WebCode	Response
4BPU69	AAC LC
4KV2K9	AAC LC
4VEDX8	Advanced Audio Coding - Low Complexity (AAC-LC)
6KCCX8	AAC LC (Advanced Audio Codec Low Complexity)
7B2AC8	Code Type: audio / mp4
7YALG8	AAC LC
8QCBL3	Quicktime: MPEG4 low complexity AAC
9PWEU3	M4A
CK6D4X	AAC LC
CN9ANZ	Advanced Audio Codec Low Complexity
CUU2YZ	Advanced Audio Codec Low Complexity (AAC LC)
EXM6GW	Mp4a or m4a container with AAC LC audio stream codec – please refer to Additional Comments section
F6YLNW	AAC-LC (Advanced Audio Codec - Low Complexity).
GW6VGV	AAC LC (Advanced Audio Codec Low Complexity)
JBVGZR	AAC LC Advanced Audio Code Low Complexity
JEUTVR	AAC LC (Advanced Audio Codec Low Complexity)
JKL37Q	AAC LC (Advance Audio Codec Low Complexity)
JRHZTT	AAC LC
JWEJVR	AAC
JXA3WT	AAC
KJNALR	m4a with advanced audio codec
LNANPN	(M4A /isom/iso2)
M84GBM	AAC LC (Advanced Audio Codec Low Complexity)
M9ELLP	AAC
MFYM4N	The format of the audio within the container is AAC LC *see additional comments
MND7MM	M4A
MQUVMQ	MPEG-4 container file: M4A (M4A /isom/iso2)
PD7AHM	MPEG-4 Audio (AAC codec)

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TABLE 1

Question 1- 2 : Examination Questions

WebCode	Response
PK6UJH	AAC-LC (Advanced Audio Codec Low Complexity)
Q29QFM	Advanced Audio Codec Low Complexity (AAC LC)
Q2LH7H	ACC
R9ZQ8K	AAC LC
T9EARH	M4A
U7PMAE	AAC
UEZPVE	single stream AAC (Low complexity), 1channel, 8kHz sample rate, data rate (max) 48kb/s
VLFPQH	AAC LC (Advanced audio Codec Low Cmplexity)
YW8DBB	AAC LC
YZ389C	AAC LC
Z2CVL9	aac

Question 1-2: What is the audio format of this MPEG-4 container file?

Consensus Result: Advanced Audio Coding (or Codec) Low Complexity or AAC LC or AAC

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TABLE 1

Question 1- 3 : Examination Questions

Question 1-3: How many audio channels are contained in the audio file? Provide a NUMERIC response.

Manufacturer's Response: 1

WebCode	Response
4BPU69	1
4KV2K9	1
4VEDX8	1
6KCCX8	1
7B2AC8	1
7YALG8	1
8QCBL3	1
9PWEU3	1
CK6D4X	1
CN9ANZ	1
CUU2YZ	1
EXM6GW	1
F6YLNW	1
GW6VGV	1
JBVGZR	1
JEUTVR	1
JKL37Q	1
JRHZTT	1
JWEJVR	1
JXA3WT	1
KJNALR	1
LNANPN	1
M84GBM	1
M9ELLP	1
MFYM4N	1
MND7MM	1
MQUVMQ	1
PD7AHM	1

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TABLE 1

Question 1- 3 : Examination Questions	
WebCode	**Response**
PK6UJH	1
Q29QFM	01
Q2LH7H	1
R9ZQ8K	1
T9EARH	1
U7PMAE	1
UEZPVE	1
VLFPQH	1
YW8DBB	1
YZ389C	1
Z2CVL9	1

Question 1-3: How many audio channels are contained in the audio file? Provide a NUMERIC response.

Consensus Result: 1

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TABLE 1

Question 1- 4 : Examination Questions

Question 1-4: What is the length in time of the audio file? Provide your answer in the following format: MM:SS.

Manufacturer's Response: 03:00

WebCode	Response
4BPU69	03:00
4KV2K9	03:00
4VEDX8	03:00
6KCCX8	03:00
7B2AC8	03:00
7YALG8	03:00
8QCBL3	03:00
9PWEU3	03:00
CK6D4X	03:00
CN9ANZ	03:00
CUU2YZ	03:00
EXM6GW	03:00
F6YLNW	03:00
GW6VGV	03:00
JBVGZR	03:00
JEUTVR	03:00
JKL37Q	03:00
JRHZTT	03:00
JWEJVR	03:00
JXA3WT	03:00
KJNALR	3 minutes 0 seconds
LNANPN	3:00.480
M84GBM	03:00
M9ELLP	03:00
MFYM4N	03:00
MND7MM	3:00.480
MQUVMQ	03 min 00 s

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TABLE 1

Question 1- 4 : Examination Questions	
WebCode	**Response**
PD7AHM	03:00
PK6UJH	03:00
Q29QFM	03:00
Q2LH7H	03:00
R9ZQ8K	03:00
T9EARH	03:00
U7PMAE	03:00.48
UEZPVE	03:00
VLFPQH	03:00
YW8DBB	03:00
YZ389C	03:00
Z2CVL9	03:00

Question 1-4: What is the length in time of the audio file? Provide your answer in the following format: MM:SS.

Consensus Result: 03:00

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TABLE 1

Question 1- 5 : Examination Questions

Question 1-5: What is the sample rate of the audio file? Provide a NUMERIC response and the associated units.

Manufacturer's Response: 8 kHz (8000 Hz)

WebCode	Response
4BPU69	Sample Rate: 8000 Units: Hz
4KV2K9	Sample Rate: 8000 Units: Hz
4VEDX8	Sample Rate: 8000 Units: Hz
6KCCX8	Sample Rate: 8000 Units: Hz (Hertz)
7B2AC8	Sample Rate: 8000 Units: Hz
7YALG8	Sample Rate: 34.7 Units: kb/s
8QCBL3	Sample Rate: 8000 Units: Hertz
9PWEU3	Sample Rate: 8000 Units: Hertz (Hz)
CK6D4X	Sample Rate: 8000 Units: Hz
CN9ANZ	Sample Rate: 8000 Units: Hz
CUU2YZ	Sample Rate: 8000 Units: Hz
EXM6GW	Sample Rate: 8000 Units: Hz
F6YLNW	Sample Rate: 8000 Units: Hz
GW6VGV	Sample Rate: 8000 Units: Hertz (Hz)
JBVGZR	Sample Rate: 8000 Units: Hz
JEUTVR	Sample Rate: 8000 Units: Hz
JKL37Q	Sample Rate: 8000 Units: Hz
JRHZTT	Sample Rate: 8000 Units: Hz
JWEJVR	Sample Rate: 8000 Units: Hz
JXA3WT	Sample Rate: 8000 Units: Hz
KJNALR	Sample Rate: 8000 Units: Hertz

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TABLE 1

Question 1- 5 : Examination Questions

WebCode	Response
LNANPN	Sample Rate: 8000 Units: Hz
M84GBM	Sample Rate: 8000 Units: Hertz (Hz)
M9ELLP	Sample Rate: 8000 Units: Hz
MFYM4N	Sample Rate: 8000 Units: Hz
MND7MM	Sample Rate: 8000 Units: Hz
MQUVMQ	Sample Rate: 8000 Units: Hz
PD7AHM	Sample Rate: 8000 Units: Hertz (Hz)
PK6UJH	Sample Rate: 8000 Units: Hertz
Q29QFM	Sample Rate: 8000 Units: Hz
Q2LH7H	Sample Rate: 8000 Units: Hz
R9ZQ8K	Sample Rate: 8000 Units: Hz
T9EARH	Sample Rate: 8000 Units: Hz
U7PMAE	Sample Rate: 8000 Units: Hz
UEZPVE	Sample Rate: 8000 Units: Hertz (Hz)
VLFPQH	Sample Rate: 8000 Units: Hz
YW8DBB	Sample Rate: 8000 Units: Hz
YZ389C	Sample Rate: 8000 Units: Hertz
Z2CVL9	Sample Rate: 8000 Units: Hz

Question 1-5: What is the sample rate of the audio file? Provide a NUMERIC response and the associated units.

Consensus Result: Sample Rate: 8000
Units: Hertz (Hz)

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TABLE 1

Question 1- 6 : Examination Questions

Question 1-6: What is the frequency value of the tonal noise present? Provide a NUMERIC response and the associated units.

Manufacturer's Response:	Frequency value/Units: 1.2 kHz (1200 Hz) Are harmonics of this tone present? No

WebCode	Response
4BPU69	Frequency Value: 1200 Units: Hz Are harmonics of this tone present?: No
4KV2K9	Frequency Value: 1200 Units: Hz Are harmonics of this tone present?: No
4VEDX8	Frequency Value: 1200 Units: Hz Are harmonics of this tone present?: No
6KCCX8	Frequency Value: 1199 Units: Hz Are harmonics of this tone present?: Yes
7B2AC8	Frequency Value: 1200 Units: Hz Are harmonics of this tone present?: No
7YALG8	Frequency Value: 8000 Units: Hz Are harmonics of this tone present?: Yes
8QCBL3	Frequency Value: 1200 Units: Hertz Are harmonics of this tone present?: No
9PWEU3	Frequency Value: 0 Units: N/A Are harmonics of this tone present?: Yes [Not within laboratory's scope.]
CK6D4X	Frequency Value: 1200 Units: Hz Are harmonics of this tone present?: Yes
CN9ANZ	Frequency Value: 1100 Units: Hz Are harmonics of this tone present?: Yes
CUU2YZ	Frequency Value: 1200 Units: Hz Are harmonics of this tone present?: No
EXM6GW	Frequency Value: 1.2 Units: kHz Are harmonics of this tone present?: No
F6YLNW	Frequency Value: 1200 Units: Hz Are harmonics of this tone present?: No
GW6VGV	Frequency Value: 1200 Units: Hertz (Hz) Are harmonics of this tone present?: No

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TABLE 1

Question 1- 6 : Examination Questions

WebCode	Response
JBVGZR	Frequency Value: 1200 Units: Hz Are harmonics of this tone present?: No
JEUTVR	Frequency Value: 1200 Units: Hz Are harmonics of this tone present?: No
JKL37Q	Frequency Value: 1199.44 Units: Hz Are harmonics of this tone present?: Yes
JRHZTT	Frequency Value: 1200 Units: Hz Are harmonics of this tone present?: No
JWEJVR	Frequency Value: 1200 Units: Hz Are harmonics of this tone present?: No
JXA3WT	Frequency Value: 1200 Units: Hz Are harmonics of this tone present?: No
KJNALR	Frequency Value: 1200 Units: Hertz Are harmonics of this tone present?: No
LNANPN	Frequency Value: 1.2 Units: Hz Are harmonics of this tone present?: No
M84GBM	Frequency Value: 1200 Units: Hz Are harmonics of this tone present?: No
M9ELLP	Frequency Value: 1200 Units: Hz Are harmonics of this tone present?: No
MFYM4N	Frequency Value: 1200 Units: Hz Are harmonics of this tone present?: No
MND7MM	Frequency Value: 1200 Units: Hz Are harmonics of this tone present?: No
MQUVMQ	Frequency Value: 205 Units: Hz Are harmonics of this tone present?: No
PD7AHM	Frequency Value: 1200 Units: Hertz (Hz) Are harmonics of this tone present?: Yes
PK6UJH	Frequency Value: 1200 Units: Hertz Are harmonics of this tone present?: No
Q29QFM	Frequency Value: 1200 Units: Hz Are harmonics of this tone present?: No

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TABLE 1

Question 1- 6 : Examination Questions	
WebCode	**Response**
Q2LH7H	Frequency Value: 1200 Units: Hz Are harmonics of this tone present?: No
R9ZQ8K	Frequency Value: 1200 Units: Hz Are harmonics of this tone present?: No
T9EARH	Frequency Value: 1200 Units: Hz Are harmonics of this tone present?: No
U7PMAE	Frequency Value: 1200 Units: Hz Are harmonics of this tone present?: No
UEZPVE	Frequency Value: 1200 Units: Hertz (Hz) Are harmonics of this tone present?: No
VLFPQH	Frequency Value: 1200 Units: Hz Are harmonics of this tone present?: No
YW8DBB	Frequency Value: 1200 Units: Hz Are harmonics of this tone present?: No
YZ389C	Frequency Value: 1200 Units: Hz Are harmonics of this tone present?: No
Z2CVL9	Frequency Value: 1200 Units: Hz Are harmonics of this tone present?: No

Question 1-6: What is the frequency value of the tonal noise present? Provide a NUMERIC response and the associated units.

<u>Consensus Result</u>: Frequency value: 1200
Units: Hertz (Hz)
Are harmonics of this tone present? No

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TABLE 1

Question 1- 7 : Examination Questions

Question 1-7: Are any clipped audio samples present in the file?

Manufacturer's Response: Yes

WebCode	Response
4BPU69	Yes
4KV2K9	No
4VEDX8	Yes
6KCCX8	Yes
7B2AC8	Yes
7YALG8	Yes
8QCBL3	Yes
9PWEU3	Yes
CK6D4X	Yes
CN9ANZ	No
CUU2YZ	Yes
EXM6GW	Yes
F6YLNW	Yes
GW6VGV	Yes
JBVGZR	Yes
JEUTVR	Yes
JKL37Q	Yes
JRHZTT	Yes
JWEJVR	Yes
JXA3WT	Yes
KJNALR	Yes
LNANPN	Yes
M84GBM	Yes
M9ELLP	No
MFYM4N	Yes
MND7MM	Yes
MQUVMQ	No
PD7AHM	Yes

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TABLE 1

Question 1- 7 : Examination Questions

WebCode	Response
PK6UJH	Yes
Q29QFM	Yes
Q2LH7H	Yes
R9ZQ8K	No
T9EARH	Yes
U7PMAE	Yes
UEZPVE	Yes
VLFPQH	No
YW8DBB	Yes
YZ389C	No
Z2CVL9	Yes

Question 1-7: Are any clipped audio samples present in the file?

Consensus Result: Yes

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

WebCode	Response
4BPU69	Continuous tone attenuation: I used the EQ module in Izoptope RX10 to apply a notch/bell filter at 1200Hz (-60dB gain), with a narrow Q (400.0). This effectively attenuated the continuous tone without negative impact on speech intelligibility. Noise reduction: This recording contains handling/rustling noise throughout, with prominent low frequency/low mid frequency energy. To reduce this, I first used EQ in Izoptope RX10 to apply a brickwall high pass filter at 100Hz and a brickwall low pass filter at 4000Hz to remove unwanted low and high frequency content. I then used the De-Wind module in RX10 to reduce some of the low frequency energy in the noise (crossover frequency set to 400Hz to reduce noise below this frequency, reduction set to 3.0 and fundamental recovery set to 5.0) and improve overall listenability. Finally, I used De-Rustle in RX10 (reduction set to 2.5) to provide further reduction of the handling noise across the full spectrum of the recording. Processed file normalization (-3 dB): After all processing was applied, i used the Normalize module in RX10 to set the target peak level (dBFS) to -3dB. Additional Clarification: Prior to applying EQ and noise reduction, I used De-Clip in RX10 to repair digitally clipped samples, with the threshold set to -0.5dB. After applying EQ and noise reduction, I used the Fab Filter Pro-L2 to reduce the difference between the louder and quieter speech. I set the attack to 67ms, the release to 264ms, with gain of 12dB applied and an output ceiling of -0.5dBTP. The file was then normalised to -3dB as the last step in the chain. Processing chain: De-Clip - repair clipped samples > EQ - HPF @100Hz, LPF@4kHz, Notch at 1200Hz to attenuate continuous tone > De-Wind - reduce low and low mid frequency energy from the noise >De-Rustle - further reduction of handling/rustling noise > Fab Filter Pro-L2 - reduce difference between louder and quieter speech > Normalize - normalise file to -3dBFS.
4KV2K9	Continuous tone attenuation: Using Audacity software, a continuous beep tone was identified. Analzye 'Plot Spectrum' to identified the spike tone frequency on '1200Hz'. Then applied a 'Notch Filter' and set Frequency(Hz) value '1200' with Q value '20'. Noise reduction: Using Audacity software, select a section containing the noise, then use 'Noise Reduction' filter and select 'Get Noise Profile' from the settings to captured the noise. Then applied 'Noise Reduction' again to the entire audio file with settings (Noise Reduction(dB)=8, Sensitivity=6, Frequency Smoothing(bands)=6). Processed file normalization (-3 dB): Using Audacity software, applied a 'Normalize' filter with settings Normalize peak amplitude to '-3.0' dB. Additional Clarification: Using Audacity software, applied a 'Filter Curve EQ', 'Normalize', 'Compressor', 'Limiter' effects to get a better speech sound.
4VEDX8	Continuous tone attenuation: Adobe Audition "Frequency Analysis" tool: Visual observation of a large peak between 1180 Hz and 1220 Hz. Found the exact peak frequency by right clicking on the Frequency Analysis graph and copying the graph data to a text file. In the resulting table there is a peak value between 1199.95 and 1200.07 Hz, rounded down to 1200 Hz. Applied a "Notch Filter" in Adobe Audition: 1200 Hz, gain -50 dB, Notch Width: Super Narrow Noise reduction: Adobe Audition: "Dynamics Processing" (expander/compressor combination) to amplify the softer sounds and reduce the level of the loud peaks. Applied a "Hard Limiter" to reduce te level of these loud peaks more. Processed file normalization (-3 dB): Adobe Audition: "Normalize (Process)" to -3 dB. Additional Clarification: All steps always involved listening to the audio before/while/after making changes.

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TABLE 2

Question 2- 1 : Enhanced Audio Examination

WebCode	Response
6KCCX8	Continuous tone attenuation: We began by working in the DC Live software, where a notch filter was applied at 1199 Hz. This adjustment lowered the intensity of that specific frequency to diminish the persistent tone while avoiding any negative impact on speech intelligibility. Noise reduction: Broadband background noise was reduced using iZotope RX 10's Voice De-noise module. The process was optimized for dialogue, with the filter set to "Surgical" mode to maintain speech integrity. A noise profile was learned from a low-speech segment, and a moderate reduction setting was applied to attenuate the steady noise floor while preserving the natural characteristics of the target speech. Processed file normalization (-3 dB): The audio file was peak-normalized using the Normalize module in iZotope RX 10. The target peak level was set to –3.0 dBFS. Additional Clarification: [Not reported by participant.]
7B2AC8	Continuous tone attenuation: The audio file was processed using (Sound Cleaner II) v1.04.944 software to enhance speech clarity and remove background noise. The file was imported into Sound Cleaner and analysed for unwanted noise such as continuous tones and ambient hiss. Noise reduction: The following filters were applied: 1. Broadband Noise Reduction: o Suppression depth: 19 dB, o Adaptation time: 0.2 s. 2. Equalizer: o Mode: Accumulated, o Scale: Logarithmic, o Band: 600-800-2048 Hz. 3. Amplifier was applied to adjust the overall signal level. Processed file normalization (-3 dB): Finally, the processed audio was normalised to –3 dB using «SIS II» Audio Forensic Software (STC-S521) v2.8.215 software and saved in uncompressed PCM WAV format, maintaining the original sample rate. Additional Clarification: Continuous tone attenuation was achieved by applying the broadband noise reduction and equaliser filters, effectively reducing steady low-frequency hum while preserving the natural characteristics of the speech. The amplifier helped optimise the signal level before normalisation, ensuring consistent output without distortion.
7YALG8	Continuous tone attenuation: Sound Remover Filter: Sound Model Complexity - 10, Sound Refinement Passes - 40, Content Complexity - 10, Content Refinement Passes - 40, Enhanced for Speech - FFT Size 4096 Noise reduction: Declipper: Gain - -1dB, Tolerance - 1%, Min. Clip Size - 3 samples, Interpolation - Cubic, 1 problem detected. 1 problem repaired. DeReverb: 70%. Hard Limiter: Peak, Maximum Amplitude - -9.0dB, Input Boost 6.0dB, Look-Ahead Time - 7ms, Release Time - 100ms. Notch Filter: Frequency - 30.0Hz, Gain - -95.1, Notch Width - Narrow, Ultra Quiet Processed file normalization (-3 dB): Normalised to -3dB before saving in .wav format. Additional Clarification: [Not reported by participant.]
8QCBL3	Continuous tone attenuation: Fast Fourier Transform (FFT) file set from 1187-1227 Hz for entire file. Noise reduction: Captured NR profiles from various points in the file that were in the proximity of conversation and applied to conversations throughout file. Amplification applied to NR. Processed file normalization (-3 dB): Applied normalization process to entire file to -3db once enhancement completed. Additional Clarification: Applied stretch/pitch (iZotope radius, set to preserve speech characteristics with a low pitch shift/semitone and pitch coherence) to certain conversations to enhance recognizable speech.

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TABLE 2

Question 2- 1 : Enhanced Audio Examination

WebCode	Response
9PWEU3	Continuous tone attenuation: The continuous tone attenuation relied on the implementation of the Notch filter to mitigate the tonal value at 1200 hertz; this was achievable via Adobe Audition's notch filter functionality,. The Notch filter allows the attenuation based on specific frequency (defined by the notch width, which was set to very narrow); this prevents the attenuation of the surrounding frequencies. I set the attenuation by -80.db to ensure the continuous tone would be eliminated. I did not enable the ultra quiet or set gain to parameters. Noise reduction: I implemented both the noise reduction and sound remover and captured both the sound print and sound model of the unwanted natural noise of the audio recording and made attempts to attenuate it. The red and yellow lines represent the the lower noise and the higher amplitude noise (commonly references as the noise floor). The green frequency parameter represents the threshold for the noise floor reduction. I set the threshold to the higher noise floor since more unwanted noises existed in that range. The additional parameters relate Reduce by and Noise Reduction; these relate to how much you want to reduce the amplification by and controlling how much noise reduction is applied to the audio. The noise reduction filter necessitates a noise print (a section of capture audio), so I attempted to capture a localized section of the audio that does not contain conversational aspects. I also implemented the sound remover filter (that utilizes a sound model learning, similar to noise print) to remove additional unwanted noise. Content complex and sound complexity relates to the convoluted noises within the audio signal based on all frequencies. The reduce by relates to the DE amplification and the refinement assists in providing a more accurate signal post processing. Processed file normalization (-3 dB): I utilized the Adobe Audition functionality to normalize the audio to -3 so the volume that was quieter would sound a bit louder while making the louder audio sound quieter Additional Clarification: The additional Clarifications derive from the recommended enhancement order of operations within the Audio Forensic Fundamentals from the Audio Engineering Society (AES); these incorporate the Parametric Equalizer, Hard Limiter, Amplify, and De-reverberation. The hard limiter greatly attenuates sound about a specified decibel; this was implemented to the preset -3 db, which also assists in hearing the lower noises based off of the input boost. The Parametric equalizer assists in attenuating frequencies based off of the low pass filter. The de-reverberation auto detects resonating noises from all frequencies and mitigates those noises based on a percentile, while amplify can boost or attenuate a signal
CK6D4X	Continuous tone attenuation: Using SIS Specialty Sound Editor (STC-S521) version 3.1.10.462, the analysis of spectrum showed a continuous tone at frequency 1200 Hz. The Noise Reduction Module in the Processing menu was used to remove the tone noise. Frame size: Auto Noise reduction: Using SIS Specialty Sound Editor (STC-S521) version 3.1.10.462, in the Noise Reduction Module in the Processing menu was used:- 1. Checkbox of Remove Wide-band Noise were enable with the following parameter: • Maximal Gain, dB : 20dB 2. Checkbox of Inverse Filter were enabled with the following parameter: • Suppressing Gain 40 dB • Amplification Gain: 25 dB • Frame Size: Auto • Automatic Speech Equalizer: Enabled Processing Mode: Entire Signal Processed file normalization (-3 dB): Using SIS Specialty Sound Editor (STC-S521) version 3.1.10.462 , after Continous Tone Attenuation and Noise Reduction steps were performs, All waveforms in the entire signal were selected, and the Normalization module under the Processing menu was used and parameter of amplitude level set to −3 dB. Additional Clarification: Using SIS Specialty Sound Editor (STC-S521) version 3.1.10.462 , the signal of characteristics the original audio file showed clipping detected at 0.05 s (0.03%). After applying Continous Tone Attenuation , Noise Reduction and Normalization steps were performs normalization to −3 dB, the signal analysis showed no clipping detected. The audio conversation became clearer after the processing

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TABLE 2

Question 2- 1 : Enhanced Audio Examination

WebCode	Response
CN9ANZ	Continuous tone attenuation: Audacity - I did a frequency analysis on a part of the audio with only the tone. I determined the tone was in a frequency range of 1150 - 1250. I used a Graphic EQ to lower that range of frequencies to zero. Noise reduction: Audacity - I did a frequency analysis on a part of the audio with only the speaker's voice. I determined the voice was a frequency around 800Hz. In parts of the audio where noise was covering the voice, I used the Noise Reduction effect to lower the volume of the noise. I then used a Graphic EQ to increase the volume of frequencies in the range of 630Hz to 1000Hz (voice). I used the Amplify effect on each segment to make the amplitude similar across the entire audio. I exported the audio to a WAV signed 16-bit PCM file. Processed file normalization (-3 dB): Audacity - I used the Normalize tool and normalized peak amplitude to -3dB. I exported the audio to a WAV signed 16-bit PCM file. Additional Clarification: There were two segments that were only very loud noise, so I muted those segments entirely.
CUU2YZ	Continuous tone attenuation: iZotope RX10: De hum: -55 dB at 1200 Hz Noise reduction: 1. iZotope RX10: De click: Sensitivity: 10, Click widening: 5 ms 2. iZotope RX10: Dialogue isolate: Dialogue gain: 4 db, Noise gain: -3 dB, Sensitivity: 10 3. iZotope RX10: Equalizer 4. Adobe Audition CS6, Noise reduction (-10 dB) Time range: 00:37.000 – 00:45.200, 00:55.000 – 00:57.900, 1:00.500 – 1:08.300, 1:26.800 – 1:28.200, 1:42.700 – 1:45.600, 1:46.700 – 1:49.800, 2:38.500 – 2:51.800 6. Processed file normalization (-3 dB): Adobe Audition CS6 (Hard limiter) Additional Clarification: -
EXM6GW	Continuous tone attenuation: 1. DeClick 2. Gain down -2 3. DeClip 4. Continuous tone attenuation: DeHum 1200 Hz 5. EQ Noise reduction: 6. Noise reduction: DeWind 7. Leveler Processed file normalization (-3 dB): 8. Processed file normalization (-3dB): Normalization to -3 dB 9. Export pcm wav file named "voicemail-2095_A5_Convert_Chain04.wav" Additional Clarification: NA
F6YLNW	Continuous tone attenuation: Applied a notch filter at 1200 Hz (Q=30) to attenuate the continuous tonal noise present throughout the recording. This removes the narrowband tone without affecting surrounding speech frequencies. Select the entire track (Ctrl +A). Go to Effect > Notch Filter… In the dialog: Frequency (Hz): 1200 Q: 30 (higher Q = narrower cut, safer for speech) Click Apply (or OK). Noise reduction: Identified a segment containing only background noise (00:09:220 - 00:09:690). Performed Noise Reduction > Get Noise Profile. Applied broadband noise reduction with: Noise reduction: 12 dB, Sensitivity: 6, Frequency Smoothing: 6 bands, Applied processing to entire file to reduce ambient noise while preserving speech. Processed file normalization (-3 dB): Select all audio again (Ctrl +A). Go to Effect > Normalize… Check Normalize peak amplitude to: -3 dB. Click OK. Additional Clarification: No dynamic range compression or EQ was applied to avoid altering the original speech characteristics beyond necessary intelligibility improvements.
GW6VGV	Continuous tone attenuation: Adobe Audition - Band-reject filter at 1200 Hz with a -40 dB attenuation. Noise reduction: Adobe Audition - Clip reduction using the DeClipper filter (normal intensity). - Intensive click reduction using the DeClicker filter (applied three times). - FFT filter applied to attenuate frequencies between 20 and 30 Hz, targeting subsonic vibrations. - Automatic click remover was applied. - Noise attenuation using selection tools and the spot healing brush. - Noise reduction set to 5% and 5 dB, with an FFT size of 2048. - Reverb cancellation applied at 30% intensity. Processed file normalization (-3 dB): Adobe Audition - Vocal enhancer filter applied to improve voice presence. - Multiband compressor used to boost vocal frequencies. - Normalization set to -3 dB to ensure consistent output levels. Additional Clarification: [Not reported by participant.]

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TABLE 2

Question 2- 1 : Enhanced Audio Examination	
WebCode	**Response**
JBVGZR	Continuous tone attenuation: The tonal noise was identified using Adobe Audition ver. 25 Frequency Analysis FFT and Spectral Frequency Display spectrograph. The m4a file was converted to a WAV PCM file format with the same sampling rate for use with the SES Omega ver. 1.0.12 forensic processing software. The WAV file was imported into the SES software to attenuate noise and increase intelligibility and clarification. I used the Tone Removal filter to attenuate the 1200 Hz tone, see filter chart below, settings in Tonal Removal section. Speech Extraction System-SES Ω v1.0.12 Copyright 1996-2025 Intelligent Devices, Inc. Status Report Main Settings { Input Mode { Sum } Output Level { 9 dB } } Engine 0 - Scene 1 - Time 0h 00m 00.000s -> END { Talker 1 { Differential { OFF } Distortion/Click { Clip Detection Threshold { 25 } Click Detection Threshold { 48 } } Hum Removal { OFF } Smooth Adaptive { OFF } Band Limiter { Low { 108 Hz } High { 4005 Hz } Attenuation { 40.0 dB } Low Enhance { BOOST } Mid Enhance { BOOST } High Enhance { OFF } } Parametric EQ { OFF } Harmonic Notch { OFF } Tone Removal { Deviation Decision Threshold { 88.4 } Adjacent Band Latching { 0 Hz } Scan Cycle Duration { 7.9 sec } Tone Rejection { 96 dB } Removed Tone List: 21.0 Hz - 42.0 Hz 1134.0 Hz - 1218.0 Hz 6132.0 Hz - 6174.0 Hz 6594.0 Hz - 6678.0 Hz 6846.0 Hz - 6867.0 Hz 7791.0 Hz - 7833.0 Hz 8904.0 Hz - 9030.0 Hz 9135.0 Hz - 9261.0 Hz 9387.0 Hz - 9408.0 Hz 9471.0 Hz - 9492.0 Hz 9555.0 Hz - 9618.0 Hz 9828.0 Hz - 9891.0 Hz 9912.0 Hz - 9933.0 Hz 10017.0 Hz - 10059.0 Hz 10101.0 Hz - 10122.0 Hz 10227.0 Hz - 10290.0 Hz 10395.0 Hz - 10458.0 Hz 10500.0 Hz - 10521.0 Hz 10584.0 Hz - 10605.0 Hz 10752.0 Hz - 10773.0 Hz 10794.0 Hz - 10857.0 Hz 11088.0 Hz - 11109.0 Hz 11130.0 Hz - 11151.0 Hz 11256.0 Hz - 11277.0 Hz 11487.0 Hz - 11550.0 Hz 11592.0 Hz - 11613.0 Hz 11907.0 Hz - 11970.0 Hz 12264.0 Hz - 12306.0 Hz 12327.0 Hz - 12369.0 Hz 12558.0 Hz - 12621.0 Hz 12936.0 Hz - 12957.0 Hz 13167.0 Hz - 13188.0 Hz 13230.0 Hz - 13293.0 Hz 13356.0 Hz - 13377.0 Hz 13419.0 Hz - 13545.0 Hz 13566.0 Hz - 13587.0 Hz 13608.0 Hz - 13629.0 Hz 13755.0 Hz - 13776.0 Hz 13860.0 Hz - 13881.0 Hz 13944.0 Hz - 13986.0 Hz 14112.0 Hz - 14133.0 Hz 14280.0 Hz - 14322.0 Hz 14343.0 Hz - 14364.0 Hz 14469.0 Hz - 14490.0 Hz 14595.0 Hz - 14658.0 Hz 14952.0 Hz - 14994.0 Hz 15036.0 Hz - 15057.0 Hz 15120.0 Hz - 15141.0 Hz 15288.0 Hz - 15309.0 Hz 15372.0 Hz - 15393.0 Hz 15624.0 Hz - 15645.0 Hz 15897.0 Hz - 15918.0 Hz 15939.0 Hz - 15981.0 Hz 16611.0 Hz - 16653.0 Hz 16968.0 Hz - 17010.0 Hz 17115.0 Hz - 17178.0 Hz 17220.0 Hz - 17346.0 Hz 17409.0 Hz - 17430.0 Hz 17472.0 Hz - 17493.0 Hz 17619.0 Hz - 17682.0 Hz 17955.0 Hz - 17997.0 Hz 18312.0 Hz - 18354.0 Hz 18480.0 Hz - 18501.0 Hz 18648.0 Hz - 18669.0 Hz 18984.0 Hz - 19005.0 Hz 19299.0 Hz - 19341.0 Hz 19551.0 Hz - 19572.0 Hz 19593.0 Hz - 19614.0 Hz 19635.0 Hz - 19698.0 Hz 19803.0 Hz - 19845.0 Hz 19908.0 Hz - 19929.0 Hz 19971.0 Hz - 20034.0 Hz 20307.0 Hz - 20370.0 Hz 20643.0 Hz - 20664.0 Hz End of Removed Tone List } Adaptive Extract { OFF } Noise Reduction { OFF } Telecom Enhance { OFF } IQ Curve Match { OFF } Voice Level Normalizer { Threshold { -15.0 } Compression { 8.0 } Compensation Gain { 0.0 } Noise reduction: SES Omega ver. 1.0.12 forensic processing software was used for noise reduction. The Bandwidth Limiter was used to reduce any noise outside the sample rate and attenuate any low frequency rumble. A Distortion/Click filter was used to repair any clipping and reduce any severe impact noises. A Voice Level Normalizer Compression Amplifier was used to further reduce the impact noise and help clarify the speech by improving the signal to noise ratio. See filter chart below, settings in Distortion/Click and Voice Level Normalizer section. Speech Extraction System-SES Ω v1.0.12 Copyright 1996-2025 Intelligent Devices, Inc. Status Report Main Settings { Input Mode { Sum } Output Level { 9 dB } } Engine 0 - Scene 1 - Time 0h 00m 00.000s -> END { Talker 1 { Differential { OFF } Distortion/Click { Clip Detection Threshold { 25 } Click Detection Threshold { 48 } } Hum Removal { OFF } Smooth Adaptive { OFF } Band Limiter { Low { 108 Hz } High { 4005 Hz } Attenuation { 40.0 dB } Low Enhance { BOOST } Mid Enhance { BOOST } High Enhance { OFF } } Parametric EQ { OFF } Harmonic Notch { OFF } Tone Removal { Deviation Decision Threshold { 88.4 } Adjacent Band Latching { 0 Hz } Scan Cycle Duration { 7.9 sec } Tone Rejection { 96 dB } Removed Tone List: 21.0 Hz - 42.0 Hz 1134.0 Hz - 1218.0 Hz 6132.0 Hz - 6174.0 Hz 6594.0 Hz - 6678.0 Hz 6846.0 Hz - 6867.0 Hz 7791.0 Hz - 7833.0 Hz 8904.0 Hz - 9030.0 Hz 9135.0 Hz - 9261.0 Hz 9387.0 Hz - 9408.0 Hz 9471.0 Hz - 9492.0 Hz 9555.0 Hz - 9618.0 Hz 9828.0 Hz - 9891.0 Hz 9912.0 Hz - 9933.0 Hz 10017.0 Hz - 10059.0 Hz 10101.0 Hz - 10122.0 Hz 10227.0 Hz - 10290.0 Hz 10395.0 Hz - 10458.0 Hz 10500.0 Hz - 10521.0 Hz 10584.0 Hz - 10605.0 Hz 10752.0 Hz - 10773.0 Hz 10794.0 Hz - 10857.0 Hz 11088.0 Hz - 11109.0 Hz 11130.0 Hz - 11151.0 Hz 11256.0 Hz - 11277.0 Hz 11487.0 Hz - 11550.0 Hz 11592.0

TABLE 2

Question 2- 1 : Enhanced Audio Examination	
WebCode	**Response**
	Hz - 11613.0 Hz 11907.0 Hz - 11970.0 Hz 12264.0 Hz - 12306.0 Hz 12327.0 Hz - 12369.0 Hz 12558.0 Hz - 12621.0 Hz 12936.0 Hz - 12957.0 Hz 13167.0 Hz - 13188.0 Hz 13230.0 Hz - 13293.0 Hz 13356.0 Hz - 13377.0 Hz 13419.0 Hz - 13545.0 Hz 13566.0 Hz - 13587.0 Hz 13608.0 Hz - 13629.0 Hz 13755.0 Hz - 13776.0 Hz 13860.0 Hz - 13881.0 Hz 13944.0 Hz - 13986.0 Hz 14112.0 Hz - 14133.0 Hz 14280.0 Hz - 14322.0 Hz 14343.0 Hz - 14364.0 Hz 14469.0 Hz - 14490.0 Hz 14595.0 Hz - 14658.0 Hz 14952.0 Hz - 14994.0 Hz 15036.0 Hz - 15057.0 Hz 15120.0 Hz - 15141.0 Hz 15288.0 Hz - 15309.0 Hz 15372.0 Hz - 15393.0 Hz 15624.0 Hz - 15645.0 Hz 15897.0 Hz - 15918.0 Hz 15939.0 Hz - 15981.0 Hz 16611.0 Hz - 16653.0 Hz 16968.0 Hz - 17010.0 Hz 17115.0 Hz - 17178.0 Hz 17220.0 Hz - 17346.0 Hz 17409.0 Hz - 17430.0 Hz 17472.0 Hz - 17493.0 Hz 17619.0 Hz - 17682.0 Hz 17955.0 Hz - 17997.0 Hz 18312.0 Hz - 18354.0 Hz 18480.0 Hz - 18501.0 Hz 18648.0 Hz - 18669.0 Hz 18984.0 Hz - 19005.0 Hz 19299.0 Hz - 19341.0 Hz 19551.0 Hz - 19572.0 Hz 19593.0 Hz - 19614.0 Hz 19635.0 Hz - 19698.0 Hz 19803.0 Hz - 19845.0 Hz 19908.0 Hz - 19929.0 Hz 19971.0 Hz - 20034.0 Hz 20307.0 Hz - 20370.0 Hz 20643.0 Hz - 20664.0 Hz End of Removed Tone List } Adaptive Extract { OFF } Noise Reduction { OFF } Telecom Enhance { OFF } IQ Curve Match { OFF } Voice Level Normalizer { Threshold { -15.0 } Compression { 8.0 } Compensation Gain { 0.0 } Processed file normalization (-3 dB): The processed audio was exported from SES Omega forensic software ver. 1.0.12 in a WAV/PCM file format with 8000Hz sample rate and 32-bit sample size. The processed audio was imported into Adobe Audition ver. 25 which was used to normalize the processed file. Using the Normalize effect from the Amplitude and Compression category of effects allowed me to type in the exact value of normalization required (-3dB). Additional Clarification: I used SES Omega ver. 1.0.12 to do additional clarification. From the Bandwidth Limiter filter I applied additional gain (+6dB) across the speech frequency range, low and middle frequencies (100Hz – 1500 Hz. This helped increase the speech levels. I used the Voice Level Normalizer Compression Amplifier filter to attenuate the louder impact noises and increase the listenability of the recording. After analyzing the recording with a Waveform monitor I was able to pinpoint the sound levels for speech and noise to properly configure the Compression Amplifier. See filter chart below, settings in the Bandwidth Limiter and Voice Level Normalizer section. Speech Extraction System-SES Ω v1.0.12 Copyright 1996-2025 Intelligent Devices, Inc. Status Report Main Settings { Input Mode { Sum } Output Level { 9 dB } } Engine 0 - Scene 1 - Time 0h 00m 00.000s -> END { Talker 1 { Differential { OFF } Distortion/Click { Clip Detection Threshold { 25 } Click Detection Threshold { 48 } } Hum Removal { OFF } Smooth Adaptive { OFF } Band Limiter { Low { 108 Hz } High { 4005 Hz } Attenuation { 40.0 dB } Low Enhance { BOOST } Mid Enhance { BOOST } High Enhance { OFF } } Parametric EQ { OFF } Harmonic Notch { OFF } Tone Removal { Deviation Decision Threshold { 88.4 } Adjacent Band Latching { 0 Hz } Scan Cycle Duration { 7.9 sec } Tone Rejection { 96 dB } Removed Tone List: 21.0 Hz - 42.0 Hz 1134.0 Hz - 1218.0 Hz 6132.0 Hz - 6174.0 Hz 6594.0 Hz - 6678.0 Hz 6846.0 Hz - 6867.0 Hz 7791.0 Hz - 7833.0 Hz 8904.0 Hz - 9030.0 Hz 9135.0 Hz - 9261.0 Hz 9387.0 Hz - 9408.0 Hz 9471.0 Hz - 9492.0 Hz 9555.0 Hz - 9618.0 Hz 9828.0 Hz - 9891.0 Hz 9912.0 Hz - 9933.0 Hz 10017.0 Hz - 10059.0 Hz 10101.0 Hz - 10122.0 Hz 10227.0 Hz - 10290.0 Hz 10395.0 Hz - 10458.0 Hz 10500.0 Hz - 10521.0 Hz 10584.0 Hz - 10605.0 Hz 10752.0 Hz - 10773.0 Hz 10794.0 Hz - 10857.0 Hz 11088.0 Hz - 11109.0 Hz 11130.0 Hz - 11151.0 Hz 11256.0 Hz - 11277.0 Hz 11487.0 Hz - 11550.0 Hz 11592.0 Hz - 11613.0 Hz 11907.0 Hz - 11970.0 Hz 12264.0 Hz - 12306.0 Hz 12327.0 Hz - 12369.0 Hz 12558.0 Hz - 12621.0 Hz 12936.0 Hz - 12957.0 Hz 13167.0 Hz - 13188.0 Hz 13230.0 Hz - 13293.0 Hz 13356.0 Hz - 13377.0 Hz 13419.0 Hz - 13545.0 Hz 13566.0 Hz - 13587.0 Hz 13608.0 Hz - 13629.0 Hz 13755.0 Hz - 13776.0 Hz 13860.0 Hz - 13881.0 Hz 13944.0 Hz - 13986.0 Hz 14112.0 Hz - 14133.0 Hz 14280.0 Hz - 14322.0 Hz 14343.0 Hz - 14364.0 Hz 14469.0 Hz - 14490.0 Hz 14595.0 Hz - 14658.0 Hz 14952.0 Hz - 14994.0 Hz 15036.0 Hz - 15057.0 Hz 15120.0 Hz - 15141.0 Hz 15288.0 Hz - 15309.0 Hz 15372.0 Hz - 15393.0 Hz 15624.0 Hz - 15645.0 Hz 15897.0 Hz - 15918.0 Hz 15939.0 Hz - 15981.0 Hz 16611.0 Hz - 16653.0 Hz 16968.0 Hz - 17010.0 Hz 17115.0 Hz - 17178.0 Hz 17220.0 Hz - 17346.0 Hz 17409.0 Hz - 17430.0 Hz 17472.0 Hz - 17493.0 Hz 17619.0 Hz - 17682.0 Hz 17955.0 Hz - 17997.0 Hz 18312.0 Hz - 18354.0 Hz 18480.0 Hz - 18501.0 Hz 18648.0 Hz - 18669.0 Hz 18984.0 Hz - 19005.0 Hz 19299.0 Hz - 19341.0 Hz 19551.0 Hz - 19572.0 Hz 19593.0 Hz - 19614.0 Hz 19635.0 Hz - 19698.0 Hz 19803.0 Hz - 19845.0 Hz 19908.0 Hz - 19929.0 Hz 19971.0 Hz - 20034.0 Hz 20307.0 Hz - 20370.0 Hz 20643.0 Hz -

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TABLE 2

Question 2- 1 : Enhanced Audio Examination

WebCode	Response
	20664.0 Hz End of Removed Tone List } Adaptive Extract { OFF } Noise Reduction { OFF } Telecom Enhance { OFF } IQ Curve Match { OFF } Voice Level Normalizer { Threshold { -15.0 } Compression { 8.0 } Compensation Gain { 0.0 }
JEUTVR	Continuous tone attenuation: RX iZotope De-hum: giving him to learn a zone of silence where there is only the tone, to remove the 1200 Hz tone. Noise reduction: RX iZotope Spectral Repair: in areas where there is noise, friction and ball bouncing, taking great care to keep the areas where voices are heard so as not to modify them excessively. RX iZotope De-crackle: to remove brief spikes from violent friction or impacts with the microphone. RX iZotope De-click: treat friction noises from clothing or the body itself that are transmitted through the microphone. Processed file normalization (-3 dB): RX iZotope Normaize: selecting Target peak level [dBFS] at -3.00 Additional Clarification: RX iZotope EQ: to equalize the frequencies where the voice is most important RX iZotope Leveler: optimize for Dialogue to adjust the levels of the different speakers where the voice fluctuates greatly in distance from the microphone or in volume.
JKL37Q	Continuous tone attenuation: full audio clip has a 1.2k hum Software used: Adobe Audition Noise reduction: Used "sound remover - effect" with preset setting and added "enhance for speech" Used "Declicker" Software used: Adobe Audition Processed file normalization (-3 dB): go to "effects", choose "amplitude & compression", choose "Normalize", type in "-3", choose dB (not percentage as already preselected) Software used: Adobe Audition Additional Clarification: N/A
JRHZTT	Continuous tone attenuation: The .m4a file was opened in Adobe Audition 2021 and saved as .wav file with the original sampling rate, channel, and bit depth. Then the .wav file was opened in "iZotope RX 10 Advanced (64-bit) v10.3.0.1775". The continuous tone was attenuated with its De-hum module with the following parameters: Learn, Filter DC offset, Filter type: Static, Frequency: 1200, Q: 224, Harmonics: 1. Noise reduction: Adobe Audition: Selected areas were normalized to 90% for vocal and 80% for some background sounds to make them clearer. iZotope RX 10 Advanced: Then Dialogue Isolate module was used to suppress some unwanted signal.Dialogue gain 0 dB, Noise gain 0 dB, Sensitivity 8.0, Ambience preservation 80, Quality (Best). Processed file normalization (-3 dB): Adobe Audition's normalization was used with parameters -3, dB, DC offset 0.0%. Additional Clarification: None.
JWEJVR	Continuous tone attenuation: iZotope RX 11 Pro > EQ band: 1200 Hz -60 dB gain with a Q of 400 and frequency precision of 3 Hz Noise reduction: all steps undertaken in iZotope RX 11 Pro 1 - marked sections with audible speech 2 - applied -30 dB gain to sections without speech 3 - applied de-rustle to sections with audible speech + rustle 4 - applied voice de-noise to whole file 5 - applied normalize to 3 dB to sections with speech Processed file normalization (-3 dB): 6 - applied normalize to 3 dB to whole file Additional Clarification: saved as WAV file with 24 bit and white noise dither

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TABLE 2

Question 2- 1 : Enhanced Audio Examination

WebCode	Response
JXA3WT	Continuous tone attenuation: Process Chain : channels : 2 sample rate : 8000.00 EQ - Linear Phase Current Settings : on/off: on LS1 frequency : 191.57, 191.57 Hz gain : -50.00, -50.00 dB slope : 120.00 dB/octave HS1 frequency : 3476.95, 3476.95 Hz gain : -50.00, -50.00 dB slope : 120.00 dB/octave Debuzz-3 Current Settings : on/off: on frequency : 500.00 Hz threshold : 0.00, 0.00 dB reduction : -96.00, -96.00 dB bandwidth : 5000.00, 5000.00 Hz detection channels : 1, 2 tracking : on harmonic mode : all Compressor Current Settings : on/off: on threshold : -1.00, -1.00 dB knee : 3.37, 3.37 dB makeup gain : 10.03, 10.03 dB read ahead : 0.00, 0.00 ms attack : 0.01, 0.01 ms hold : 20.00, 20.00 ms release : 333.00, 333.00 ms ratio : 11.84, 11.84 parallel : -50.00, -50.00 dB link : on link channels : 1, 2 side chain Vintage Decrackle Current Settings : on/off: on mode : crackle 2 detect : on split level : 37.33, 37.33 dB auto split : off threshold : 40.00, 40.00 FNR Current Settings : on/off: on speed : 0.20, 0.20 Hz resolution : optimal bias : 0.00, 0.00 dB attenuation : -5.53, -5.53 dB focus : 82.00, 82.00 DNS Two Current Settings : on/off: on threshold bias : 0.00, 0.00 dB attenuation : -2.73, -2.73 dB learn : false SUMMARY: Track Loudness View Standard : ATSC A/85 Loudness : -19.54 LKFS True Peak : 1.55 dB TP Compliant : false Out of range : Loudness ,True Peak Noise reduction: see above Processed file normalization (-3 dB): Adobe Audition Normalization -3 dB Additional Clarification: see above
KJNALR	Continuous tone attenuation: The Adobe Audition notch filter was applied to reduce the tonal noise at 1200 Hz. Noise reduction: The Adobe Audition 30 band graphic equalizer was applied to reduce broadband noise below 200 Hz and above 5000 Hz. Processed file normalization (-3 dB): The Adobe Audition normalize process tool was applied to normalize the audio at -3 dB. Additional Clarification: Adobe Audition version 25.2.0.123 was used to clarify the audio recording. The audio was processed in this order: 1) Speech Leveler tool to balance the sound of speech. 2) The notch filter was applied. 3) The 30 band EQ tool was applied. 4) The normalization tool was applied. 5) The clarified audio was saved as an uncompressed wav file.
LNANPN	Continuous tone attenuation: Audition - applied the Notch filter for 1.2k noise Noise reduction: Audition - applied noise reduction 60% and reduced by 20dB for effective tone attenuation Processed file normalization (-3 dB): Audition - Normalize to -3dB to ensure the audio maintains a consistence volume level with out distortion Additional Clarification: The audio was reported to have been recorded from a cellular phone in the POIs pocket. Due to the motion of the phone in that pocket, there is significant interference as the microphone appears to be rubbing on the pocket fabric. This had a negative effect on the efficacy of the enhancement and prevents a full enhancement of all speech.
M84GBM	Continuous tone attenuation: -Adobe Audition, Notch filter (frequency 1200 Hz, Gain -50dB) Noise reduction: -Adobe Audition, Parametric equalizer (high pass filter; 250 Hz, Gain 24dB/Oct) Processed file normalization (-3 dB): -Adobe Audition, Normalize (-3dB) Additional Clarification: -Adobe Audition, iZotope RX10 plugin De-clip (high quality, -1dB) -Adobe Audition, Dynamic processing; to increase quiet speech (between -20 and -40 dB) -Adobe Audition Parametric equalizer; increased higher voice frequencies to clarify speech (frequency 2800hz, Gain 7dB)
M9ELLP	Continuous tone attenuation: Audio file was uploaded into OTExpert 6.0, under the Edit > Filter option, used Target filtering and select the range 1180Hz to 1210Hz. Set the frequency to -10dB to reduce the continuous tone attenuation. Noise reduction: Under Edit > Filter > Subtract noise. Set the values as Range: 50 to 400Hz, out of range: -3dB with Change noise energy of 0.1 and substract spectrum. Processed file normalization (-3 dB): Under Edit > Normalization, tick only the Normalize and set to 70% of maximum which is equivalent to -3dB. Additional Clarification: [Not reported by participant.]

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TABLE 2

Question 2- 1 : Enhanced Audio Examination

WebCode	Response
MFYM4N	Continuous tone attenuation: Izotope EQ band filter at 1200 Hz Noise reduction: Izotope Normalization to -0.70, De-click, De-Rustle, Voice De-noise adaptive mode, De-plosive modules Processed file normalization (-3 dB): Izotope Normalization Additional Clarification: Izotope Leveler module
MND7MM	Continuous tone attenuation: A continuous tone was observed at 1200Hz throughout the recording. This was attenuated using Izotope RX10 Pro EQ module with a notch filter applied at 1200Hz (Q40, -60 dB gain). This process reduced the interfering tone while preserving intelligibility. Noise reduction: A. De-click Algorithm: Multi-band (Random Clicks) Settings: Sensitivity 8.6, Bias 0.9 Purpose: Removed sharp transient spikes caused by basketball impacts and friction bursts. Result: Smoother waveform with fewer distracting high-energy peaks. B. De-crackle Mode: High Quality Strength: 8.3 Purpose: Reduced low-level grain and fine crackle generated by fabric rubbing. Result: Produced a cleaner background with reduced constant static noise. C. Equalization (EQ) Mode: Analog EQ Adjustments: - High Shelf @ 200 Hz (+0.19 dB) – restored mild low-end warmth. - Notch Filter @ 1200 Hz (–60 dB, Q = 400) – removed persistent hum. - Low Roll-off @ ≈62 Hz – attenuated sub-bass from basketball thumps. Result: The hum was fully eliminated, and the vocal range became clearer and more balanced. D. Clip Gain Automation Method: Manual node editing (multiple add/move operations). Purpose: Evened out the inconsistent amplitude caused by the movement of the recording device against the fabric. Result: Achieved consistent loudness across speech regions without compressing natural dynamics. E. De-rustle (Multiple Passes) in isolated areas Settings: Strength 1.6–6.3; Ambience Preservation 43–81. Purpose: Incrementally reduced broadband rustle while retaining natural background presence. Result: improvement in clarity, with rustling reduced to a minimal level. Processed file normalization (-3 dB): Final Step: Normalized to –3 dB peak using Izotope RX10 Pro as per test specification. Purpose: Ensured consistent output level for playback while preserving headroom. Result: Standardized overall loudness, completing the enhancement chain. Additional Clarification: Workflow Sequence 1. Critical Listening and Assessment - Identified eight clipped points (non-speech) — no de-clipping required. - Noted primary noise sources: rustle, hum, and impact transients. 2. File Conversion (Audacity – Lossless WAV Export) 3. De-click and De-crackle for transient and background smoothing. 4. EQ Correction to remove 1200 Hz hum and balance frequencies. 5. Clip Gain Automation for manual dynamic control. 6. Multiple De-rustle Passes for broadband noise reduction in selected areas 7. Normalization to –3 dB for final level alignment.

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TABLE 2

Question 2- 1 : Enhanced Audio Examination	
WebCode	**Response**
MQUVMQ	Continuous tone attenuation: The wind noise in voicemail-2095(audio) was very high, so I used the MyEdit Wind Remover tool to reduce it. After reducing the wind noise, the resulting voicemail-2095(audio) file was resampled to a normal sampling rate, and all further processing was carried out at this rate. To reduce noise and increase the intelligibility of voicemail-2095(audio), I have used different filters of the Gold Wave and Audacity tools; the filters used in this action were: 1. A high-pass filter was applied to remove low-frequency noise and improve the clarity of the voicemail-2095(audio). The procedure was as follows: open Gold Wave software > import the audio file (File > Open) > select all. Go to Effect > Filter > Low/Pass Filter and apply the following settings: initial cutoff at 70 Hz, final cutoff at 90 Hz, high-pass mode with dynamic enabled, steepness set to 12, and the preset of Fade Out Bass > ok. 2. Noise reduction was applied to minimize unwanted sounds such as hum/buzz and other background disturbances, thereby improving the clarity and intelligibility of the voicemail-2095(audio). The procedure was as follows: open Gold Wave > import the audio file (File > Open) > highlight each section containing unwanted noise at the following time intervals: 21.940–22.071 s, 39.152–39.290 s, 43.872–45.146 s, 55.405–57.831 s, 1:03.945–1:03.817 s, 1:26.015–1:29.009, 1:47.027–1:51.077 s, 1:57.152–1:57.392 s, 2:28.087–2:28.372 s, 2:37.001–2:40.002 s, 2:40.988–2:46.506 s, and 2:48.036–2:52.887 s. For each selected segment, go to Effect > Filter > Noise Reduction > choose the preset Hiss Removal (-50 dB), and apply the following settings: FFT size of 12, overlap of 4x, and scale of 100%. Afterward, I selected the entire audio file and go to Effect > Filter > Noise Reduction. I choose to Reduce Hum, then applied the following settings: FFT size 12, overlap 4x, and 100% scale. 3. An equalizer filter was used to adjust the levels of specific frequency ranges, allowing targeted enhancement or reduction of low, mid, and high frequencies in voicemail-2095(audio). This used to minimize unwanted noise without affecting the rest of the recording and improved the overall tonal balance. The procedure was as follows: open Gold Wave > import the audio file (File > Open), > select all. Go to Effect > Filter > Equalizer, choose the preset (Equal Loudness), and apply the automated Equal Loudness settings to complete the process. 4. A low-pass filter was applied to remove high-frequency noise during the audio enhancement process. This filter effectively reduced the persistent hiss in the recording and improved intelligibility by attenuating high-frequency noise that was masking the desired speech. The procedure was as follows: open Gold Wave > import the audio file > select the entire audio > go to Effects > Filter > Low/High-Pass Filter. Choose Low-Pass, select the preset (Muffled), and apply the automated settings. Muffled as preset (this automatically sets cutoff frequency and steepness) > Click OK. Software used: Gold Wave, Filter used: Low-Pass Filter (LPF), Mode used: Static, Cutoff Frequency: 800 Hz, Steepness: 10, Preset: Muffled. Noise reduction: Noise reduction was applied to minimize unwanted sounds such as hum/buzz and other background disturbances, thereby improving the clarity and intelligibility of the voicemail-2095(audio). The procedure was as follows: open Gold Wave > import the audio file (File > Open) > highlight each section containing unwanted noise at the following time intervals: 21.940–22.071 s, 39.152–39.290 s, 43.872–45.146 s, 55.405–57.831 s, 1:03.945–1:03.817 s, 1:26.015–1:29.009, 1:47.027–1:51.077 s, 1:57.152–1:57.392 s, 2:28.087–2:28.372 s, 2:37.001–2:40.002 s, 2:40.988–2:46.506 s, and 2:48.036–2:52.887 s. For each selected segment, go to Effect > Filter > Noise Reduction > choose the preset Hiss Removal (-50 dB), and apply the following settings: FFT size of 12, overlap of 4x, and scale of 100%. Afterward, I selected the entire audio file and go to Effect > Filter > Noise Reduction. I choose to Reduce Hum, then applied the following settings: FFT size 12, overlap 4x, and 100% scale. Processed file normalization (-3 dB): To normalize voicemail-2095(audio) to –3 dB, I used Audacity. Normalization adjusts the overall amplitude of the recording so that it reaches a clear, and professional loudness level without changing its natural dynamics. This process ensures a balanced output, improves speech intelligibility, maintains consistent volume, and helps to prevent the distortion. The procedure was as follows: Open Audacity > import the audio file (File > Open) > select all > go to Effect > Volume and Compression > Normalize > set the peak amplitude to –3 dB > apply. After normalization, the audio was exported using the following procedures, File > Export Audio > choose export location and file name > Export normalized audio. Additional Clarification: I downloaded the file 25-5591_Audio.zip from the CTS Portal and saved it to the computer for integrity verification. I calculated the MD5 and SHA1 hash values of the downloaded file (MD5: 819bbd94bc743b096860246e6b47dc25; SHA1: cb104af3b372c1abf5195ceb07607c95730983d3) and compared them with the reference values of

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TABLE 2

Question 2- 1 : Enhanced Audio Examination

WebCode	Response
	hash value that were provided by CTS, message digest five (MD5): 819bbd94bc743b096860246e6b47dc25; and the hash secure algorithm one (SHA1): cb104af3b372c1abf5195ceb07607c95730983d3. The matching results confirmed that the downloaded file was intact and unaltered. After verifying the file's integrity, I proceeded to extract the contents of the ZIP archive. The file contained inside: voicemail-2095(audio), audio assessment made using the media info. format: PCM, codec ID:1, duration: 3min0sec, bit rate:128kb/s, channel:1, sampling rate:8000Hz, bit depth:16bits. The listenability of this voicemail-2095 was very low, caused by different signs that were understandable as wind noise (very high), tones(hiss or Hum/buzz),etc.... Second, the wind noise in voicemail-2095 was very high, so I used the MyEdit Wind Remover tool to reduce it. After reducing the wind noise, the resulting voicemail-2095 file was resampled to a normal sampling rate, and all further processing was carried out at this rate.
PD7AHM	Continuous tone attenuation: A Notch Filter was applied at 1200 Hz using the DC live. The gain at 1200 Hz was reduced to weaken the continuous tone without making the speech harder to understand. Noise reduction: A 10 Band Graphic EQ was applied to reduce the background noise in the audio, we lowered the low frequencies (31-125 Hz) and raised the middle frequencies (500-2000 Hz) so the speech becomes clearer and we kept the high frequencies (8000-1600 Hz) low to reduce the hiss , after adjusting the levels we applied the EQ to the audio. Processed file normalization (-3 dB): The processed audio was normalized to a peak level of -3 dBs using the Normalize function in DC live to ensure consistent playback volume while preventing clipping. Additional Clarification: [Not reported by participant.]
PK6UJH	Continuous tone attenuation: iZotope RX 10 EQ module: notch consisting of -60 dB at 1200 Hz, bell with Q=400. Noise reduction: iZotope RX 10 De-click to reduce transients in masking noise: Algorithm = Multi-band periodic clicks; Frequency skew = 6.1; Sensitivity = 6.6; Click widening = 3 ms iZotope RX 10 De-wind to reduce lower frequencies in masking noise: Reduction = 2.7; Crossover frequency = 1500 Hz; Fundamental recovery = 5.0; Artefact smoothing = 10.0 iZotope RX 10 De-ess to reduce higher frequencies in masking noise: Algorithm = Spectral; Threshold = -9.5 dBFS; Cuttoff freq = 800 Hz; Speed = fast; Spectral Shaping = 11%; Spectral tilt = +53 iZotope RX 10 EQ to reduce low frequency noise: High pass brick wall filter at 130 Hz Zynaptiq Unfilter: adaptive filter to reduce convolution on speech and increase intelligibility by flattening frequency response. Processed file normalization (-3 dB): iZotope RX 10 Loudness Control: True peak = -3.0 dBTP; Integrated loudness = - 18 LKFS; Tolerance = 2.0 LU; program loudness gate = on Additional Clarification: iZotope RX 10 Leveler was used to increase level of voices relative to masking noise peaks: Optimise for dialogue; Target level = -20 dB FS; Responsiveness = 0.6; Preserve dynamics = 0 Replay of processed WAV file was confirmed on standard [Laboratory] Service Windows PC using VLC media player.

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TABLE 2

Question 2- 1 : Enhanced Audio Examination

WebCode	Response
Q29QFM	Continuous tone attenuation: Tonal Noise Removal: The file was imported into Audacity software. To identify the frequency value of the tonal noise present and presence of any harmonics, Plot spectrum feature of audacity was used. Only tonal noise area was selected at different selection points in audio file and spectrum was obtained using Plot spectrum. A sharp tonal noise at 1200 Hz was identified by selecting different areas where only tonal noise was present. After that whole audio was selected, and Notch filter was applied at 1200 Hz with Q=50. Notch Filter Settings: Frequency: 1200 Hz Q: 50 (Screenshots in case folder) Manual Selective Editing: To remove remaining tonal noise at the start of audio, amplification to -50 dB was applied on that small tonal segment only. Noise reduction: Noise Reduction Filter: A section of the recording containing just pocket friction/ rubbing/ masking noise was selected and the noise profile was generated using Noise Reduction filter "Get Noise Profile". After that all the whole audio file was selected and Noise was reduced with following parameters. Noise Reduction Filter Noise reduction: 12 dB Sensitivity: 3 Frequency Smoothing: 8 bands High-Pass Filter: Cut below 80Hz Roll-off/Slope: 12 dB/oct is applied in High pass filter to reduce low-frequency basketball thumps without removing them. To reduce Remove rumble and low-frequency noise. Legacy Compressor: (To Smooths speech levels before manual enhancement/ boosting. Even out speech peaks without raising noise too much) Threshold: -35 dB Noise Floor: -50 dB Ratio: 3:1 Attack: 0.1 sec Release: 3 sec or at most 5sec Make-up gain for 0 dB after compressing (Selected) Compress based on Peaks (Deselected) Manual Selective Editing: (To Boost low voices and suppress noise without damaging speech.) Speech segments Only: Speech/ voice segments only are amplified using 0.75dB, +3dB to +10dB on vocal/ voice selections depending upon voice levels while avoiding clipping for raise voice levels Noise-only segments: Amplify -10 dB to -15dB (to suppress noise only regions). (Screenshots saved at each step). Graphic EQ Filter: (To clarify speech intelligibility) 2kHz–4 kHz frequency range is boosted by +2 dB Processed file normalization (-3 dB): Normalization: Normalize peak amplitude: -3dB Remove Dc offset (Selected). The clarified/ enhanced audio is saved in the uncompressed PCM Wave format, maintaining the original file sample rate. Additional Clarification: The work flow is provided in order of enhancement steps. Clarifications performed are mentioned in Noise Reduction Section.
Q2LH7H	Continuous tone attenuation: Adobe Audition 23.3.0.55: Graphic Equalizer (30 bands), attenuate 1.25kHz tone Noise reduction: Adobe Audition 23.3.0.55: Parametric Equalizer, Generic Hi-Pass Preset, frequency 20Hz set to -38.5dB gain and 112Hz set to -17.2 dB to attenuate rustling noise. Processed file normalization (-3 dB): Saved the processed file out as instructed with same 8000Hz sample rate as PCM WAV file. Reimported to Adobe Audition 23.3.0.55, Normalize to -3dB as instructed, then saved as an 8000Hz sample rate as PCM WAV file. Additional Clarification: Adobe Audition 23.3.0.55: Declicker - 44 of 45 clicks repaired. Adobe Audition 23.3.0.55: Speech Volume Leveler, Soft preset.
R9ZQ8K	Continuous tone attenuation: Select the time segment from 00:03.645 to 00:04.500 as the noise sample, and perform noise reduction on the entire recording. Noise reduction: Use the vocal enhancement function to enhance the male voice. Processed file normalization (-3 dB): Apply the normalization function with the parameter set to -3 dB. Additional Clarification: Locate the segments with suspected basketball sounds and increase the gain by 10 dB.

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TABLE 2

Question 2- 1 : Enhanced Audio Examination

WebCode	Response
T9EARH	Continuous tone attenuation: Tool used: iZotope RX10 Advanced Audio Editor v10.5.0.2330 Filter used: Spectral Repair: Attenuate Frequency Range to Apply: 1150 to 12500 Hz Parameters set: Bands: 1024, Multi-resolution: ON, Surrounding region length (%): 50, Before/after weighting: 0.0, Strength: 1.0, Direction of interpolation: Vertical Noise reduction: Tool used: iZotope RX10 Advanced Audio Editor v10.5.0.2330 Filter used: EQ Parameters set: High Pass 48 dB/oct 70 Hz Freq, Low Pass 48 dB/oct 2600 Hz Freq Filter used: De-rustle Parameters set: Reduction strength: 5.5, Ambiance preservation (%): 75, Separation algorithm: Advanced joint-channel Filter used: De-wind Parameters set: Reduction: 4, Crossover frequency (Hz): 1000, Fundamental recovery: 0.0, Artifact smoothing: 5.0 Filter used: De-click Parameters set: Algorithm: Multi-band (random clicks), Sensitivity: 8.5, Frequency skew: -4.0, Click widening (ms): 3.0 Filter used: Spectral De-noise Parameters set: Adaptive mode: ON, Learn time (s): 2.5, Threshold Noisy: 0.0, Threshold Tonal: 1.0, Reduction Noisy: 19.0, Reduction Tonal: 14.0, Quality: C, Artifact control: 7.0, Smoothing: 0.0 Algorithm Behavior Smoothing: 7.0, Algorithm Behavior Algorithm: Extreme, Algorithm Behavior Multi-resolution: ON, Noise Floor Synthesis: 0.0, Noise Floor Masking: 10.0, Noise Floor Enhancement: 5.0, Noise Floor Whitening: 4.0, Dynamics Knee: 1.5, Dynamics Release (ms): 80 Filter used: Leveler Parameters set: Optimize for: Dialogue, Target level: -30.7, Responsiveness: 1.5, Preserve dynamics: 35, Ess reduction (dB): OFF, Breath control (dB): OFF Export: Format: WAV, Bit depth: 32-bit (float), BWF: OFF Processed file normalization (-3 dB): Tool used: iZotope RX10 Advanced Audio Editor v10.5.0.2330 Filter used: Normalize Parameters set: Target peak level (dBFS): -3.00 Export: Format: WAV, Bit depth: 32-bit (float), BWF: OFF Additional Clarification: To transcode the original file for further clarification: Tool used: ffmpeg version 7.1.1, parameters set: -c:a pcm_s32le
U7PMAE	Continuous tone attenuation: Rewrapped M4A as PCM WAV retaining sample rate using a batch file containing the following FFMPEG script. for %%a in ("*.m4a") do ffmpeg -i %%a -acodec pcm_s16le %%~na_PCM.wav pause Created rewrapped file voicemail-2095_PCM.wav Opened voicemail-2095_PCM.wav in IZotope RX9 and listened to recording. The audio contains what sounds like a conversation between two individuals. Intelligibility of speech is poor and compression artefacts appear to be present. Speech levels vary and are often obscured by rustling and other environmental noises including a consistent percussive noise. Broadband background noise is low however the recording is marred by a continuous tone measured at 1200Hz at approximately -37dBFS in level. During the recording, the conversations are interspersed with what sounds like straining or grunting sounds which may indicate individuals involved in physical activity. Ran Waveform Stats across the whole recording: True Peak Level: +0.34dB Possible clipped samples: 8 DC Offset: -0.010% Applied De-Clip to repair clipping artefacts present on recording with threshold (dB) set to 7.0 and -7.0 and makeup gain of -3dB Applied De-Hum to remove DC Offset Ran Waveform Stats to check True Peal level -0.80dB Possible clipped samples: 0 DC Offset: 0.000% Continuous tone attenuation was achieved by applying Spectral Repair and cutting 1200Hz tone to -62dBFs Noise reduction: Applied filters such as De-Rustle and De-Click in sections where speech was heard to attemp to improve intelligibility and reduce harshness to improve listenability. File saved as voicemail-2095_PCM_Working.wav Opened voicemail-2095_PCM_Working.wav in Adobe Audition to attempt to reduce dynamic range. Applied Compressor settings: Threshold: -34.9dB Ratio: 4:1 Attack: 1ms Release: 304.1ms Makeup Gain 17dB Processed file normalization (-3 dB): Applied Peak Normalisation to -3dB using Adobe Audition Normalisation filter. Additional Clarification: After applying De-Rustle and De-Click, applied EQ with HPF from 150Hz 12dB/Oct, 2.4dB boost at 1992Hz (1.4Q) and 5.8dB boost at 4kHz (2.1Q) to increase presence.

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TABLE 2

Question 2- 1 : Enhanced Audio Examination

WebCode	Response
UEZPVE	Continuous tone attenuation: Reduced 1200Hz tone using a high-Q notch filter (IIR). -50dB, Q=80 Noise reduction: Used static parametric EQ (FIR) to address low frequency noise, and mid frequency resonances including: Low shelf cut (210Hz), presence boost (3200Hz), and multiple notches to remove resonances (-6dB @ 336Hz, 677Hz, 585Hz). Used a compressor / Limiter to reduce dynamic range between target speech and high level device handling transients. (-16dB Threshold; 10:1 ratio; 0.1ms attack, 50ms release) Processed file normalization (-3 dB): Cedar 'Peak Normaliser' set to -3dB (sample peak). 'True Peak' (inter-sample) estimated to be -0.96dB. Additional Clarification: Cedar Cambridge signal path: File Replay > DC Filter > EQ-P(IIR -tone attenuation) > EQ-L(FIR - EQ noise reduction & clarification) > Compressor > Adaptive Limiter (-3dB) > Peak Normaliser > WAV File result
VLFPQH	Continuous tone attenuation: izotope RX 10 Noise reduction: Spectral repair, Ce-click, Ce-crackle, dialogue isolate Processed file normalization (-3 dB): Normalisation 0 Additional Clarification: [Not reported by participant.]
YW8DBB	Continuous tone attenuation: iZotope RX Advanced 10.5.0 - Conversion of audio to WAV LPCM, 24 bit, sample rate maintained - De-hum. Notch Filter @ 1200 Hz, Q of 3000. Noise reduction: iZotope RX Advanced 10.5.0 - De-Clip at -1 dBFS - De-click, sensitivity 3.0, no widening - High pass filter @ 100 Hz, 48 db/oct slope - Spectral Inverse Filter (Spectral denoise). Used samples where only static noise present - Dewind, crossover freq 640 Hz, reduction 4 - Leveler. Target -12.2. Responsiveness 1 second - Spectral Inverse Filter (Spectral denoise). Used samples where only static noise present - Transient noise attenuation - Manual gain, -48 dB in regions containing no speech - Main gain, +6 dB in regions containing low amplitude speech Processed file normalization (-3 dB): iZotope RX Advanced 10.5.0 Normalisation -3 dB Additional Clarification: A/B comparisons throughout A/B comparison at the end using Adobe Premiere Pro 25.5
YZ389C	Continuous tone attenuation: I used Izotope RX 10 program: - chose 1200 Hz and cut it off Noise reduction: Izotope RX 10: - using both spectral repair and spectral denoise (learn command) to attenuate all those parts of the signal that has disturbing sounds (one by one), expect those parts that have speech under the noise Processed file normalization (-3 dB): Izotope RX 10: - normalization -3 dB Additional Clarification: [Not reported by participant.]

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TABLE 2

Question 2- 1 : Enhanced Audio Examination

WebCode	Response
Z2CVL9	Continuous tone attenuation: In Cedar Cambridge, I corrected waveform issues (clipping, discontinuous waveforms) and applied noise reduction and EQ. Noise reduction: In Cedar Cambridge, attenuated the body and envelope of the noise transients using adaptive filters. Then applied layers of noise reduction that targeted specific activity across the frequency spectrum. Processed file normalization (-3 dB): In Cedar Cambridge, I attenuated dynamic range using a spectral dynamic limiter and compressors. I exported the file into iZotope RX7 and adjusted the loudness to be consistent with the Audio Engineering Society (AES) [Country] region Loudness guidelines. I then normalised the file to -3dB as requested by this proficiency test. Additional Clarification: The signal processing chain was as follows: 1. Import a working copy from a created E01 of provided files to Cedar Cambridge. 2. Correct clipping. 3. Correct discontinuous waveforms. 4. Attenuate continuous tone and surrounding frequencies with noise reduction. 5. High pass, transient attenuation and further continuous tone reduction with EQ. 6. Adaptive noise filters to remove body of noise. 7. Dynamic spectral limiter to tame harshness of noise transients. 8. Layered noise reduction targeting specific frequency ranges of the "noise". 9. Compression to tame the dynamic range and improve listenability of audio file. 10. Export the file from Cedar Cambridge using the following settings: pcm_s24le, 8000 Hz, 24 bit. 11. Import the file exported from Cedar Cambridge into iZotope RX7. 12. Correct waveform phase to allow accurate loudness measurements. 13. Apply target loudness settings. 14. Normalize to -3dB. 15. Export completed audio file using the following settings: pcm_s24le, 8000 Hz, 24 bit. Refer to the Cedar Cambridge report and iZotope run sheet for the filters/settings applied.

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 expected.

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Forensic Audio Enhancement Observations

TABLE 3

Part 2: Audio Enhancement Instructions -
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.
C. Normalize processed file to -3 dB. It is expected that no speech data will be degraded or lost due to the filtering process.

WebCode	Observational Notes
4BPU69	Submitted enhanced file(s) were reviewed by an expert who confirmed that this participant completed all requested audio enhancement steps.
4KV2K9	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 audio sample at 00:46.908 has a maximum sample peak level of -0.97 dB, and many other samples exceed -3 dB as well.
4VEDX8	Submitted enhanced file(s) were reviewed by an expert who confirmed that this participant completed all requested audio enhancement steps.
6KCCX8	Submitted enhanced file(s) were reviewed by an expert who confirmed that this participant completed all requested audio enhancement steps.
7B2AC8	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 maximum audio signal amplitude is 0 dB with two clipped samples.
7YALG8	Submitted enhanced file(s) were reviewed by an expert who confirmed that this participant completed all requested audio enhancement steps.
8QCBL3	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 audio is normalized to -3 dB as requested. However, while the continuous tonal noise at 1,200 Hz was removed, the notch filter settings were too wide/deep. All frequency information from approx. 1,080 Hz to 1,240 Hz was removed, including speech data.
9PWEU3	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 A observational note: While noise was reduced, over-filtration resulted in speech sounding overly rhythmic with unnatural/abrupt variations in loudness of speech. Step C observational notes: The audio is normalized to -3 dB as requested. However, low-level speech was removed by the filtering processes. Some portions of intelligible speech in the original recording were made unintelligible by over filtering (e.g., De-reverberation).
CK6D4X	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 audio sample at 00:44.607 has a maximum sample peak level of -0.78 dB, and other samples exceed -3 dB as well.
CN9ANZ	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 A observational note: Noise was reduced, however speech intelligibility was degraded. Step C observational note: The audio is normalized to -3 dB as requested. However, while the continuous tonal noise at 1,200 Hz was removed, the EQ filter removed almost all frequency information from approximately 750 Hz to 1,700 Hz, including significant speech data. Additionally, "click" artifacts from another filtering process were added throughout the recording.
CUU2YZ	Submitted enhanced file(s) were reviewed by an expert who confirmed that this participant completed all requested audio enhancement steps.
EXM6GW	Submitted enhanced file(s) were reviewed by an expert who confirmed that this participant completed all requested audio enhancement steps.
F6YLNW	Submitted enhanced file(s) were reviewed by an expert who confirmed that this participant completed all requested audio enhancement steps.

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TABLE 3

Forensic Audio Enhancement Observations	
WebCode	**Observational Notes**
GW6VGV	Submitted enhanced file(s) were reviewed by an expert who confirmed that this participant completed all requested audio enhancement steps.
JBVGZR	Submitted enhanced file(s) were reviewed by an expert who confirmed that this participant completed all requested audio enhancement steps.
JEUTVR	Submitted enhanced file(s) were reviewed by an expert who confirmed that this participant completed all requested audio enhancement steps.
JKL37Q	Submitted enhanced file(s) were reviewed by an expert who confirmed that this participant completed all requested audio enhancement steps.
JRHZTT	Submitted enhanced file(s) were reviewed by an expert who confirmed that this participant completed all requested audio enhancement steps.
JWEJVR	Submitted enhanced file(s) were reviewed by an expert who confirmed that this participant completed all requested audio enhancement steps.
JXA3WT	Submitted enhanced file(s) were reviewed by an expert who confirmed that this participant completed all requested audio enhancement steps.
KJNALR	Submitted enhanced file(s) were reviewed by an expert who confirmed that this participant completed all requested audio enhancement steps.
LNANPN	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 maximum audio signal amplitude was 0 dB with over 50 clipped samples.
M84GBM	Submitted enhanced file(s) were reviewed by an expert who confirmed that this participant completed all requested audio enhancement steps.
M9ELLP	Submitted enhanced file(s) were reviewed by an expert who confirmed that this participant completed all requested audio enhancement steps.
MFYM4N	Submitted enhanced file(s) were reviewed by an expert who confirmed that this participant completed all requested audio enhancement steps with the following notes. A narrower Q (bandwidth) of the EQ filter avoids affecting a wider range of frequencies to attenuate just the tone.
MND7MM	Submitted enhanced file(s) were reviewed by an expert who confirmed that this participant completed all requested audio enhancement steps.
MQUVMQ	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 B observational note: File was saved as PCM Wave but sample rate was changed to 16,000 Hz. Step C observational note: The maximum audio signal amplitude is 0 dB with multiple samples exceeding -3 dB.
PD7AHM	Submitted enhanced file(s) were reviewed by an expert who confirmed that this participant completed all requested audio enhancement steps.
PK6UJH	Submitted enhanced file(s) were reviewed by an expert who confirmed that this participant completed all requested audio enhancement steps.
Q29QFM	Submitted enhanced file(s) were reviewed by an expert who confirmed that this participant completed all requested audio enhancement steps.
Q2LH7H	Submitted enhanced file(s) were reviewed by an expert who confirmed that this participant completed all requested audio enhancement steps with the following notes. The volume of the continuous tone was reduced, but it is still audible.
R9ZQ8K	Submitted enhanced file(s) were reviewed by an expert who confirmed that this participant completed all requested audio enhancement steps.
T9EARH	Submitted enhanced file(s) were reviewed by an expert who confirmed that this participant completed all requested audio enhancement steps.
U7PMAE	Submitted enhanced file(s) were reviewed by an expert who confirmed that this participant completed all requested audio enhancement steps.

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TABLE 3

Forensic Audio Enhancement Observations	
WebCode	**Observational Notes**
UEZPVE	Submitted enhanced file(s) were reviewed by an expert who confirmed that this participant completed all requested audio enhancement steps.
VLFPQH	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 B observational note: File was saved as PCM Wave but sample rate was changed to 16,000 Hz. Step C observational note: The maximum audio signal amplitude is 0 dB.
YW8DBB	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 audio is normalized to -3 dB as requested. However, low-level speech was removed during the filtering process. The notes do not indicate what filter(s)/settings were applied via the iZotope RX Spectral Repair tool.
YZ389C	Submitted enhanced file(s) were reviewed by an expert who confirmed that this participant completed all requested audio enhancement steps.
Z2CVL9	Submitted enhanced file(s) were reviewed by an expert who confirmed that this participant completed all requested audio enhancement steps.

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Additional Comments

TABLE 4

WebCode	Additional Comments
4BPU69	Prior to applying processing in RX10, I made a PCM wav copy of the file using FFMPEG v6.0 with sample rate at 44.1kHz at 24 bit. Following processing, i used FFMPEG to downsample the file to 8000Hz as requested (bit depth kept at 24 bit). All processing reviewed against original unprocessed file before return. Overall I think processing has helped improve the listenability of the recording, effectively reducing the prominent tone and handling noise. There are sections of intelligible speech but further improvements to speech intelligibility have not been possible due to the low sample rate and codec attributes of the recording. I have also had this processing peer reviewed by another competent practitioner, with the following comments: "Assessed processing. A suitable approach with an improvement to listenability, mostly through the reduction of the 1.2kHz sine tone. Further processing is limited by the low-quality codec used. Experimented by swapping around the order of the De-Wind and EQ processes - no audible difference. No suggestions for improvement. I believe that the processing has been pushed as far as possible without having a detrimental effect on the intelligibility of speech."
6KCCX8	The enhancement process was performed conservatively to preserve the integrity of the original speech characteristics. All filtering steps were limited to attenuation of tonal and broadband noise without altering the linguistic content, phonetic cues, or temporal structure of the recording. No portions of the audio were removed, replaced, or time-modified. All observations regarding the presence of additional background speech or activity are based solely on audible content revealed after noise reduction, and not on any interpretive or reconstructive processing.
7B2AC8	Forensic analysis of the voicemail recording revealed clear sounds of a bouncing basketball, shoe movement, and outdoor reverberation. These acoustic features are consistent with activity on a basketball court. Therefore, the findings indicate that the person of interest was playing a one-on-one basketball game during the period in which he claimed to be home alone, contradicting his alibi.
8QCBL3	Mono signals are particularly challenging to enhance. Once enhanced, the file appeared to feature either multiple players, or at least verbally active participants, commenting on the game and interacting throughout. This did not sound like a simple one-on-one game in a public space.
9PWEU3	I did not answer Question 1-6 since it was outside the laboratory scope of examination. I filled in answers to the questions since it's a requirement. I put the number 0 to satisfy answering the question
EXM6GW	Question 1-2 is not clear on whether it is asking for the format of the container or the format of the audio stream inside the container (containers can have multiple streams such as video, audio, and subtitles). If the question had been phrased “what is the codec or audio encoding scene used in the MPEG-4 container file” the answer would have been AAC. However, it asked for the “audio format” and that could be interpreted as Mp4a/m4a or AAC. Sometimes multimedia definitions such as container, format, file type, extension, etc. are used interchangeably or inconsistently. Question 1-7 is difficult to definitively answer, because the Waveform Statistics in iZotope RX indicate “Possibly clipped samples” and the number is very low at “8.” A different audio tool, Adobe Audition, reported that there are only 2 possibly clipped samples. Digital metering can be unreliable with such short durations (less than one thousandths of a second at 8 kHz sample rate) and it is hard to say if measurement uncertainty is the cause, or if there truly is clipping.
JBVGZR	I don’t usually use an additional application to normalize audio files but since the request was so specific I used one that could be set numerically. I also did not use adaptive filters that can sometimes attenuate small amounts of speech since the request required that no speech data will be degraded or lost.
JEUTVR	Since the file in mpeg 4 format gave problems when opened with the version of the RX iZotope software available in this Laboratory, before starting the improvement process, it was converted to PCM format using the code: ffmpeg -i voicemail-2095.m4a -c:a pcm_s16le voicemail-2095_ffmpeg.wav from the FFMPEG library.
MFYM4N	* Question 1-2 was unclear. As written, the question appears to refer to the container itself while the more logical subject of the question would refer to the audio within the container.
PD7AHM	The original file voicemail-2095.m4a was processed in DC live A notch Filter at 1200 Hz was applied to reduce the continuous tone. A 10 Band Graphic EQ was applied to reduce the background noise in the audio; The final audio was normalized to –3 dB and exported as a WAV file. All speech content was

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TABLE 4

WebCode	Additional Comments
	preserved throughout the enhancement process. The improved quality allowed clearer identification of the background activity. It was determined that the individual in the recording was walking while speaking, and there were additional voices indicating interaction with other persons present at the scene.
PK6UJH	Downloaded ZIP archive was copied to laboratory workstation and SHA-1 hash was verified with QuickHash software before target M4A file was extracted. Some dynamics processing was required to reduce the level disparity between the masking noises and the speech signal, thus the iZotope RX 10 Leveler and Loudness Control modules were used to this effect.
Q29QFM	The “Part 2: Audio Enhancement Instructions” were followed as provided by CTS, and according to section 4 of the SWGDE Best Practices for the Enhancement of Digital Audio, and an enhanced/ clarified audio file was exported. (Screenshots) were saved at each step. One zip file is provided containing enhanced audio file. Hash Value of Enhanced file: "Enhanced voicemail-2095.wav" is: MD5: 3aea91f8fc73497de3b19ed0b04009b1
U7PMAE	Little improvement was made to the overall intelligibility of the recording due to the apparent concealment of the recording device and rustling noises that marred the recording. Some sections of the recording were made clearer by removing rustling and clicks. At 1.17.6s (621517 samples) and 2:27s (1180173 samples), phrases sound clearer and may be of investigative value. At 30.399s (243197 samples) a sound like a dog barking is present. At 2:27.5s (1180173 samples) a percussive impact sound can be heard. Please Note: During the zip file verification process, I noticed an anomaly between the SHA1 and MD5 hash ids. The hash strings I generated from the zip file are not the same as those listed. I raised this issue with my Quality Unit who contacted CTS. CTS maintained the hash IDs are correct, but I am unable to get a match despite using four applications to verify, all with the same results. (See uploaded email chain). [Email chain not included in this report.]
Z2CVL9	Processed audio file was a "working copy" copied from an E01 created of the provided USB 3.0 flash drive. All file copy and transfers were MD5 or SHA-256 hash verified using NUIX Evidence Mover or TeraCopy.

-End of Report-

(Appendix may follow)

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Collaborative Testing Services ~ Forensic Testing Program

Test No. 25-5591: Forensic Audio Analysis

DATA MUST BE SUBMITTED BY **Nov. 24, 2025, 11:59 p.m. EST** TO BE INCLUDED IN THE REPORT

Participant Code: U1234A WebCode: FDWDWQ

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 person of interest (POI) claims that he was home alone during a specific time window relevant to an ongoing investigation. However, a voicemail stored on a third party's phone reveals that the POI's cellular phone, during that same period, inadvertently placed a call from a pocket during a one-on-one basketball game at a public court.

Investigators would like to examine the voicemail recording to determine whether any speech or background sounds captured during the call contradict the individual's stated alibi. Audio clarification processing is needed to improve listenability and increase intelligibility through noise reduction and speech clarification.

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.
25-5591_Audio.zip MD5 hash value: 819bbd94bc743b096860246e6b47dc25
25-5591_Audio.zip SHA1 hash value: cb104af3b372c1abf5195ceb07607c95730983d3

Part 1: Examination Questions

1-1). What is the SHA-256 hash value of the voicemail-2095.m4a file?

1-2). What is the audio format of this MPEG-4 container file?

1-3). How many audio channels are contained in the audio file? Provide a NUMERIC response.

1-4). What is the length in time of the audio file? Provide your answer in the following format: MM:SS.

1-5). What is the sample rate of the audio file? Provide a NUMERIC response and the associated units.

Sample Rate:

Units:

1-6). What is the frequency value of the tonal noise present? Provide a NUMERIC response and the associated units.

Frequency Value:

Units:

Are harmonics of this tone present?

Yes ○ No ○

1-7). Are any clipped audio samples present in the file?

Yes ○ No ○

Part 2: Audio Enhancement Instructions

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.
C. Normalize processed file to -3 dB. It is expected that no speech data will be degraded or lost due to the filtering process.

Uploaded file name:

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

Continuous tone attenuation:

Noise reduction:

Processed file normalization (-3 dB):

Additional clarification:

Additional Comments

Note: *Please use appropriate punctuation to indicate the end of sentences, sections, and statements in the free-form space below. Extra spacing and returns used for separation within your text will not transfer and may cause your information to be illegible in the Summary Report. The use of lists and tabular formats to deliver information is also cautioned against, as these do not transfer.*

Optional Questions

1: List software tools used to complete this test?

2: How long did this test take to complete?

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)

____