Technical Procedure for Audio Clarification

1.0 Purpose - The purpose of this procedure is to clarify the audio using a digital copy of the original audio.

2.0 Scope - This procedure describes the steps to be taken in forensic clarification of audio during analysis by personnel in the State Crime Laboratory.

3.0 Definitions – N/A

4.0 Equipment, Materials and Reagents

- Playback device
- Audio clarification and filtering device (e.g., PCAP)

5.0 Procedure

5.1 Connect the selected playback device to the audio clarification and filtering device. This connection must be made with the highest quality connection available to both devices.

5.2 Set the system bandwidth of the audio clarification and filtering device. Several factors shall be considered when determining the system bandwidth:

5.2.1 Human speech range: Normal human speech occurs between 200-5000 Hz. Frequencies beyond this range do not increase intelligibility.

5.2.2 Media limitations: Different types of media have different limitations when it comes to the frequency range supported. See the Digital Audio Processing Training Manual from Digital Audio Corporation for more information. Also, the manufacturer of the media is a good resource for information about the media.

5.2.3 Equipment limitations: The equipment used to transmit or record the original audio can also affect the frequency range supported. For example, telephone conversations are limited to 3200 Hz by the equipment that the telephone company uses to transmit the audio.

5.3 Determine what, if any, filters or processes would possibly clarify the audio.

5.3.4 Sample Rate Correction: Occasionally, when a digital audio file is copied to CD as a standard CD audio track, the sample rate conversion required for this process will be done incorrectly. An incorrect sample rate conversion will introduce distortion into the recording. This can be identified and corrected using a non-linear editor (e.g., Adobe Audition).

5.3.5 Highpass Filter: This filter reduces audio below a certain frequency. It is typically used to reduce low frequency rumble, such as engine noise or clothing rubbing against a microphone.

5.3.6 Lowpass Filter: This filter reduces audio above a certain frequency. It is typically used to reduce high frequency hiss, such as wind noise.

5.3.7 Notch Filter: This filter reduces audio at a specific frequency. It is typically used to reduce noise from interfering hum tones that remain constant, such as a 60 Hz hum from an electrical source.

5.3.8 Comb & Multiple Notch Filters: These filters reduce audio at multiple specific frequencies. They are typically used to reduce noise from interfering hums or tones that occur at multiple frequencies.

5.3.9 One Channel (1CH) Adaptive Filter: This filter reduces repetitive and predictable noises. It is typically used to reduce noise from machines, varying tones, reverberations, and echoes.

5.3.10 Two Channel (2CH) Adaptive Filter: This filter reduces repetitive and predictable noises from a reference source. It is typically used to reduce music or TV noise for which a reference is available.

5.3.11 Spectral Inverse Filter (SIF): This filter reduces repetitive and predictable noises. It is typically used to reduce noise from machines and varying hums and tones.

5.3.12 Adaptive Spectral Inverse Filter (ASIF): This filter reduces repetitive and predictable noises. It is typically used to reduce noise from machines, varying tones, reverberations, and echoes.

5.3.13 Noise Reducer: This filter reduces random noises that are difficult to predict. It is typically used to reduce clicks, pops, wind noise, and other random noises.

5.3.14 Graphic Equalizer: This type of filter is used to adjust the audio to a more standard voice spectrum. This typically increases speech intelligibility.

5.3.15 Limiter/Compressor/Expander (LCE): This process increases or decreases gain based on the current level of the audio. It is typically used in near-party/far-party situations where one voice is inaudible, but another voice can be heard well.

5.3.16 Automatic Gain Control (AGC): This process increases gain based on the current level of the audio and is typically a final process to increase the overall output level.

5.4 The settings for all filters and/or processes shall be documented in the case notes.

5.5 Standards and Controls – Standard 1 kHz test tone on the media type submitted as evidence.

5.6 Calibrations - The hardware and software used in casework shall be verified before each case to ensure that they are functioning properly. The procedure for this verification process can be found in the Audio Performance Verification Procedure.

5.7 Maintenance – N/A

5.8 Sampling – N/A

5.9 Calculations – N/A

5.10 Uncertainty of Measurement – N/A

6.0 Limitations - Failure to attempt all relevant, available clarification filters and/or processes when analyzing audio could result in missed information that may be pertinent to a case. If excessive, unwarranted filters and/or processes are used during the analysis, the final audio may contain artifacts resulting from over-processing.

7.0 Safety – N/A

8.0 References

- Digital Audio Corporation – Digital Audio Processing Training Manual
- Audio Performance Verification Procedure

9.0 Records – N/A

10.0 Attachments – N/A

Revision History		
Effective Date	Version Number	Reason
09/17/2012	1	Original Document