

frontline®



Audio Expert System

Reference & User Guide

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Chapter 1 About this Guide

This guide is divided into two parts:

- Frontline Audio Expert System™ Reference guide and
- Audio Expert System™ User Guide.

The Reference Guide, Chapters 2, is a thorough description of Frontline's Bluetooth® audio analysis features, terminology, event types, and audio measurements.

The User Guide, Chapter 3, provides specific instructions for using the Audio Expert System™ as an integral component of the Frontline software.

Users new to *Bluetooth* audio analysis will find the Reference Guide chapter useful reading prior to using the Audio Expert System™ for troubleshooting *Bluetooth* audio. Experienced users may want to skip to the User Guide Chapter, but they will find the Reference Guide chapter a useful refresher or a quick reference when troubleshooting *Bluetooth* audio problems.

All users will find the User Guide beneficial as it describes all of the Audio Expert System™ functions, buttons, controls, views, and options in detail. The User Guide information is also found in the ComProbe software help, in the [Frontline Soderia User Manual](#) (PDF format), [Frontline BPA 600 User Manual](#) (PDF format), the [Frontline Soderia On-Line Help](#), and [Frontline BPA 600 On-Line Help](#). These last two documents are available at FTE.com.

1.1 What You Will Need

The Audio Expert System™ operates in conjunction with the Frontline software installed on your computer and with a Frontline BPA 600 Dual Mode *Bluetooth* Protocol Analyzer or Frontline Soderia Wideband *Bluetooth* Protocol Analyzer. The Frontline analyzer must be licensed for operation with the Audio Expert System™.

To use the Audio Expert System™ you will need

- Audio Expert System™ license. The license is purchased from Frontline Sales that can be contacted at sales@fte.com or 1-800-359-8570 or +1-434-984-4500.
- Frontline software installed on your computer. The software release must be after 1 August, 2014.

- Frontline BPA 600 or Frontline Sodera with an active license. The Frontline analyzer must be connected to your computer to use the Audio Expert System™ in either live capture or captured file mode.

Chapter 2 Audio Expert System™ Reference Guide

2.1 Audio Expert System™ Overview

The primary goal of the Audio Expert System™ is to expedite the detection and resolution of Bluetooth protocol related audio impairments. To achieve this, the system automatically identifies audio impairments and reports them to a user as “events”. It also correlates the audio events with any detected codec or *Bluetooth* protocol anomalies (events). The system allows a user to view the audio waveform, audio events, codec events, and *Bluetooth* protocol events on a time-aligned display.

The system works in two modes of operation:

- Non-Referenced Mode: The system is processing audio of completely unknown program content (see [Non-Referenced Mode Event Reporting on page 3](#)).
- Referenced Mode: The system is operating in a “pseudo closed loop” test scenario where the user plays specific Reference Audio files (pre-recorded audio test files provided by Frontline) on the Source DUT (device under test) that is sent via Bluetooth to the Sink DUT. Frontline’s Audio Expert System™ analyzes resulting over-the-air audio (see [Referenced Mode Event Reporting on page 7](#)).

The Audio Event Processor is a subsystem of Frontline’s Audio Expert System™, and it monitors and analyzes Bluetooth audio streams with the purpose of detecting and reporting audio impairments.

2.1.1 Non-Referenced Mode Event Reporting

In Non-Referenced Mode, the system is typically processing audio of completely unknown program content (e.g. arbitrary music or speech content). Since the system does not have any prior knowledge of the audio being analyzed, the types of audio analysis that can be performed is limited.

The following events are reported whenever the system is operating in Non-Reference mode. These are the meaningful audio analysis that the system can perform without reporting too many false positive results.

1. Volume Level (Low Volume or High Volume) [Warning]: Reported if the average volume level is not in a range conducive to performing meaningful audio analysis.
2. Clipping [Error]: Amplitude distortion due to a signal amplitude exceeding the maximum value that can be represented by the digital system.

3. Dropout [Warning]: Abrupt and very short duration intervals of silence.
4. Glitch [Error]: Extremely large sample-to-sample audio amplitude transitions that have little probability of occurring within natural speech or music.

2.1.1.1 Low Volume Alarm Event

The purpose of the Low Volume Alarm event is to warn the user that the volume level of the detected audio is below the level needed for performing meaningful audio analysis. This event is only reported in Non-Referenced Mode.

The system will not start reporting Low Volume Alarm events until a volume level above the “Low Volume Threshold” (see Figure 2.1) has been detected. After this condition is satisfied, a Low Volume Alarm event is generated when the detected peak volume drops below the Low Volume thresholds listed below for 10 consecutive 0.5 sec measurement intervals (i.e. 5 sec total). This event will not be repeated until the detected audio volume once again rises above the Low Volume Threshold and then falls below this threshold again for 10 more consecutive 0.5 sec measurement intervals.

- Speech Low Volume Threshold: -15 dBFS
- Music Low Volume Threshold: -20 dBFS

Note that if the volume level for any audio sample exceeds the Low Volume Threshold during any 0.5 sec measurement period, the Low Volume Alarm state machine resets and begins looking again.

The information reported with the Low Volume Alarm event is listed below:

- Info1: Average (RMS) level just prior to alarm event. This is the average level over the last 0.5 sec interval just prior to issuing the alarm.
- Info2: The time at when the system began detecting low volume (i.e. 5 sec before the event was reported)
- Msg: Not used.

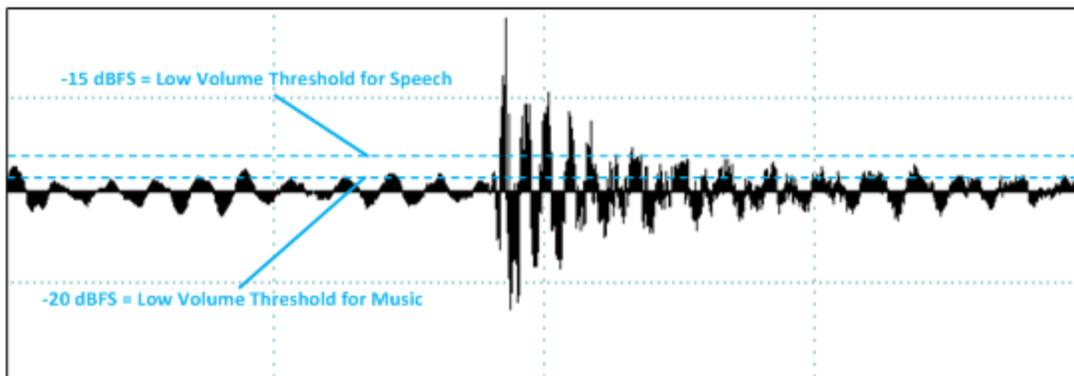


Figure 2.1 - Low Volume Threshold Example

2.1.1.2 Clipping Event

The purpose of the Clipping event is to report the presence of probable distortion that occurs when the amplitude of a signal exceeds a digital system's ability to represent it accurately. Clipping is a type of amplitude distortion.

The system reports a Clipping event when consecutive samples at the maximum value that can be represented by the digital system have been detected. Note that the maximum value that can be represented is different

depending on the number of bits per sample (i.e. bits of resolution) of the audio stream. The system limits the number of reported clipping events to typically 10 to 20 per second.

The number of consecutive samples needed to qualify as a clipping event depends on both sample rate and number of bits per sample. [Table 2.1 - Clipping Event Threshold](#) specifies the number of consecutive samples at the maximum value level that will generate a Clipping event.

Table 2.1 - Clipping Event Threshold

Consecutive Samples	Sample Rate, samples/sec	Resolution, bits
3	8000	16
5	16000	16
11	41000	16
2	64000	16
12	48000	16
24	96000	16

The information reported with the Clipping Event is listed below:

- Info1: contains an integer value representing the RMS average audio level when the clipping event was detected. The value can be converted to a decibel level via the relationship

$$20\text{Log}_{10} \left(\frac{\text{Info1}}{32767} \right).$$

- Info2: Not used.
- Msg: Not used.

2.1.1.3 High Volume Alarm Event

The purpose of the High Volume Alarm event is to warn the user that the volume level of the detected audio is above the maximum acceptable level for performing meaningful audio analysis (i.e. approaching a level where the audio will likely become distorted). This event is only reported in Non-Referenced Mode.

A High Volume Alarm event is generated when the detected audio volume is continuously above the high volume threshold (see Figure 2.2) for 10 consecutive 0.5 sec measurement intervals (i.e. 5 sec total). The event will not be repeated again until the detected volume level drops below the high volume threshold. The High Volume Alarm event will then be reported again if the high volume condition returns for 10 more consecutive 0.5 sec measurement intervals.

Listed below are the high volume threshold levels. Note that the thresholds are different for music versus speech since the crest factor (difference between peak and average levels – see [Crest Factor on page 21](#)) is different for those types of audio. Note that speech is typically transported in A2DP connections, and music is typically transported in SCO connections.

- Speech: greater than -6 dBFS
- Music: greater than -12 dBFS

The info reported with the High Volume Alarm event is listed below:

- Info1: Average (RMS) level at the start of the 5 sec interval
- Info2: The time at when the system began detecting high volume (i.e. 5 sec before the event was reported)
- Msg: Not used.

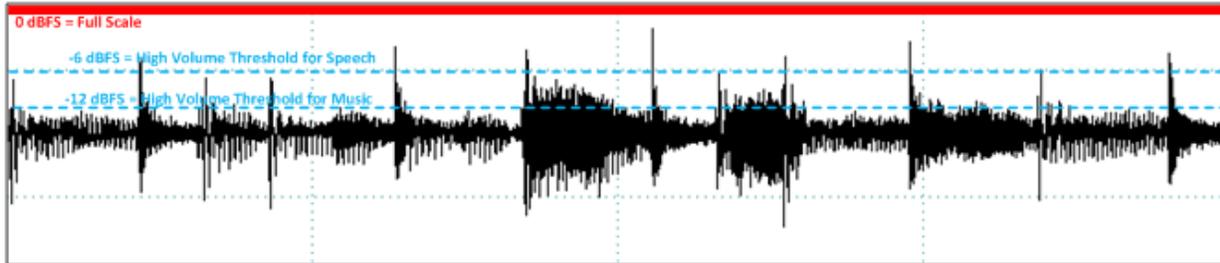


Figure 2.2 - High Volume Threshold Example

2.1.1.4 Dropout Event

The purpose of the Dropout event is to report the detection of an unusual brief silence period where the brief silence is preceded and followed by “normal” audio levels. A typical definition of Dropout is the short dramatic loss of volume typically caused by lost digital information. Root causes include transmission system errors resulting in lost data packets, transmission channel reconfigurations, bad sections of memory, processor overloads that temporarily interrupt the flow of information, and so on. The challenge is to detect real dropout events while at the same time honoring natural silence. For music, there are natural musical “rests” that occur. Speech includes natural inter-syllable, inter-word, and pause silence.

Dropout events are reported when the average audio level (RMS) is initially above the Measurement Threshold, then falls below the Silence Threshold, and then quickly rises above the Measurement Threshold again (see Figure 2.3). This approach largely disqualifies the natural inter-syllable silence and pauses that occur in natural speech, but will detect gaps caused by dropped data. Note that the system does not report dropouts that begin at very low energy levels.

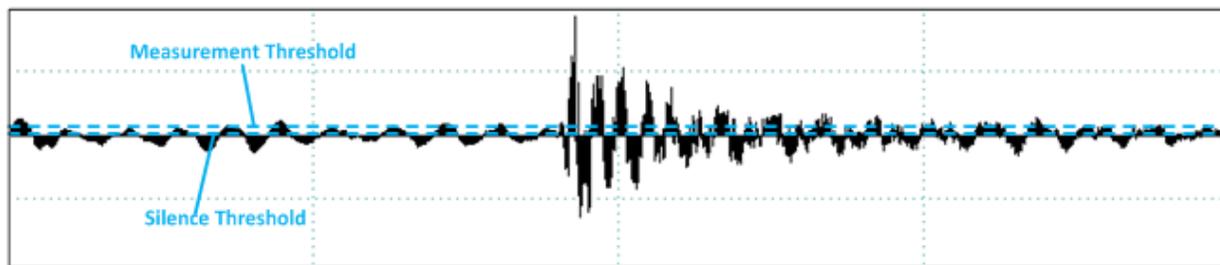


Figure 2.3 - Measurement And Silence Threshold

Shown in Figure 2.4 are examples to illustrate the Dropout event.

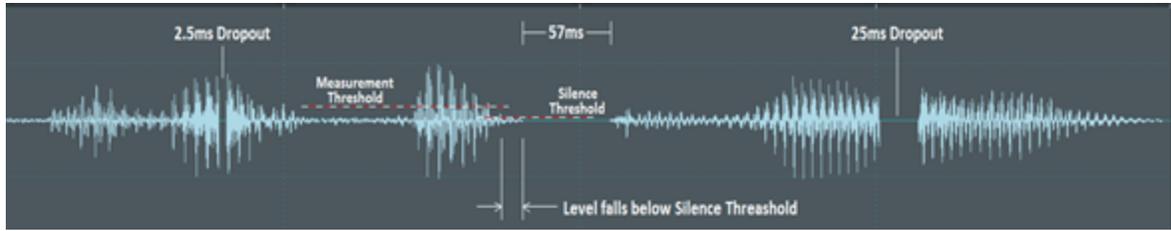


Figure 2.4 - Dropout Examples

The two events of 2.5 ms and 25 ms are reported as Dropout events, while the 57 ms interval at the center is not. Although 57 ms interval is of sufficient duration, and the level remains below Silence Threshold for the duration, the interval is not immediately preceded by sustained energy above the Measurement Threshold.

The info reported with the Dropout event is listed below:

- Info1: The number of samples over which the dropout persisted. The value can be converted to time duration (sec) by dividing by sample rate. Likewise the Start Time of the dropout can be calculate by subtracting the duration from the event's sample number (time stamp) parameter and converting to time.
- Info2: Not used.
- Msg: Not used.

2.1.1.5 Glitch Event

Glitch events are extremely large sample-to-sample audio amplitude transitions that have little probability of occurring within natural speech or music. Such dramatic changes would typically happen only in situations of dropped samples.

The system uses different thresholds for sample-to-sample transitions for music versus speech as shown below.

- Speech: greater than 40 dB change from sample-to-sample
- Music: greater than 90 dB change from sample-to-sample

The info reported with the Glitch event is listed below:

- Info1: An integer value representing the absolute value of the sample delta. This can be converted to a change in decibels by the relationship

$$20\text{Log}_{10}(\text{Info1}) .$$

- Info2: Not used.
- Msg: Not used.

2.1.2 Referenced Mode Event Reporting

In Referenced Mode, the system operates in a “pseudo closed loop” test scenario where the user plays specific Reference Audio files (pre-recorded audio test files provided by Frontline) on the Source DUT (Device Under Test). See [Referenced Mode Processing on page 18](#) for a more complete description. The events listed in the following sections are reported when the system is operating in Referenced Mode.

The following events are reported whenever the system is operating in Reference mode.

1. Test ID Found Event [Information]: Reported when a valid Test ID has been recognized.
2. Test Script Not Found Event [Error]: Reported when a valid Test ID was found but the system cannot locate the script for that Test ID.
3. Invalid Test Script Event [Error]: Reported when an error occurs while the system is verifying information in a test script.
4. Synchronization Lost Event [Error]: Reported when the system is in Referenced Mode and encounters unexpected frequencies or durations of audio segments while analyzing a received Reference Audio file.
5. Unexpected Frequency Event [Error]: Reported when a measured frequency deviates from an expected frequency by a specific percentage.
6. Unexpected Level Event [Error]: Reported when the measured level at the start of a tone segment is not within tolerance.
7. Unexpected Duration Event [Error]: Reported when a tone segment of the Reference Audio file is shorter or longer than expected.
8. Amplitude Fluctuations Event [Warning]: Reported if the system detects unexpected amplitude changes over a given interval.
9. Clipping Event [Error]: Reported when consecutive samples at the maximum value that can be represented by the digital system have been detected.
10. Unexpected Phase Change Event [Error]: Reported if the system detects a phase change during programmed intervals of constant frequency.
11. Excess Noise Event [Error]: Reported when energy sufficiently above the “Silence Threshold” is detected during programmed segments of silence.
12. CVSD HF Level Too High Event [Error]: Reported when a CVSD encoded audio stream is detected and there is high frequency energy above 4 kHz that is greater than -20 dBFS.
13. End of Test Event [Information]: Reported to indicate that the system has completed processing a test script for a Reference Audio file, and that the system has exited Reference Mode.

2.1.2.1 Test ID Found Event

The Test ID Found event (see [Test ID on page 18](#)) occurs when a valid Test ID has been recognized. A valid Test ID must meet the level, frequency, duration, and delimiter requirements. If any of these parameters do not match, the process is terminated and is reset to the initial conditions.

Note: Until a Test ID is successfully recognized, the system will continue to operate in Non-Referenced Mode; therefore, no events related to false starts are reported. This is because for arbitrary audio there is no expectation of any Test ID.

The info reported with the Test ID Found event is listed below:

- Info1: The Test ID
- Info2: The measured RMS average audio of the 400 Hz terminator tone. This level can be converted to dBFS by the relationship

$$\text{dBFS} = 20 \text{Log}_{10} \left(\frac{\text{Info1}}{32767} \right)$$

Optionally the value can be converted to “Channel Gain” via the relationship

$$\text{dB} = 20 \text{Log}_{10} \left(\frac{\text{Info1}}{23196} \right)$$

- Msg: The measured frequency [Hz] of the terminator tone.

2.1.2.2 Test Script Not Found Event

This event occurs if a valid Test ID was found (see [Transition to Referenced Mode on page 18](#)), but the script for that Test ID was not found. The system reverts to Non-Referenced Mode if this happens. This event should not occur if valid Reference Audio files provided by Frontline are being used.

The information reported with the Test Script Not Found event is listed below:

- Info1: The Test ID
- Info2: The peak level (amplitude) of the recognized Test ID terminator tone (dB).
- Msg: A message indicating the exact measure frequency of Test ID Terminator tone (Hz).

2.1.2.3 Invalid Test Script Event

This event is generated when an error occurs while accessing information in a script. This event should not occur if using a valid reference audio file provided by Frontline.

The information reported with the Invalid Test Script event is listed below:

- Info1: The Test ID
- Info2: For most cases this is the audio segment number at which the error occurred. For the special case that initialization fails, Info2 holds the number of segments found.
- Msg: Contains a case specific message indicating the nature of the message.

2.1.2.4 Synchronization Lost Event

This event is generated when the system is in Referenced Mode after a successful initial Test ID recognition (see [Synchronization on page 19](#)) and encounters unexpected frequencies or durations of audio segments while analyzing a received Reference Audio file. If this situation occurs, the internal segment tracking logic attempts to look forward and/or backward in the test script to determine if the currently measured characteristics are consistent with the previous or next segment of the script. If there is a match, the internal segment pointer is advanced or retarded appropriately, the Synchronization Lost event is not generated, and the audio analysis continues. However, if a match cannot be found, the system declares itself out of sync and generates the Synchronization Lost Event, terminates any active test script, and reverts to Non-Referenced Mode.

The information reported with the Synchronization Lost event is listed below:

- Info1: An integer reason code such as 12=Unexpected Frequency.
- Info2: Unused.
- Msg: Contains a diagnostic message containing a code point indicator, the SegmentID of the script that was expected when the error was declared, and the zero based index into the Segment Array.

2.1.2.5 Unexpected Frequency Event

In Referenced Mode, the system knows the Reference Audio file that is being played on the Source DUT; therefore, the system knows which frequencies (tones) to expect at a given time. The Unexpected Frequency event is reported when a measured frequency deviates from an expected frequency by a specific percentage (determined by the negotiated parameters of the over-the-air audio stream).

The information reported with the Unexpected Frequency event is listed below:

- Info1: The measured frequency (Hz).
- Info2: The expected frequency (Hz).
- Msg: “SegID=N”, where N is a Segment Identifier indicating the expected segment.

2.1.2.6 Unexpected Level Event

In Referenced Mode, the system knows the Reference Audio file that is being played on the Source DUT; therefore, the system knows which amplitude level to expect at a given time. The Unexpected Level event is reported when the measured level at the start of a tone segment is not within tolerance. The tolerance is dependent on sample rate and bits per sample, but it generally is +/- 3 dB for speech and +/-11 dB for music.

Since the exact transmitted amplitude levels from the Source DUT depend on a number automatic and manual analog and digital gain settings, the measured levels are normalized before comparing against limits. See [2.6 Audio Volume/Level Calibration on page 25](#).

The information reported with the Unexpected Level event is listed below:

- Info1: The measured normalized level (dB).
- Info2: The expected level (dB).
- Msg: A string containing the “Normalization Gain” factor (dB).

2.1.2.7 Unexpected Duration Event

The Unexpected Duration (UD) event is reported when a tone segment of the Reference Audio file is shorter or longer than expected. In Referenced Mode, the system knows the Reference Audio file that is being played on the Source DUT and therefore knows how long a specific tone segment should last. If either a change of amplitude or frequency arrives either before or after that programmed duration, then the change is by definition unexpected. Lost or corrupted data, repeated data, faulty packet-loss-concealment algorithms, etc can cause this type of audio impairment.

The amount that a measured duration must deviate from the programmed duration of a tone segment before the system declares this event varies, depending upon the negotiated over-the-air audio stream specific parameters, but it is generally in the range of 5% to 10%. Note that this event will result in an attempt to resynchronize (see [Synchronization Lost Event on page 9](#) and [Synchronization on page 19](#)) if the measured duration is greater than expected.

The info reported with the Unexpected Duration event is listed below:

- Info1: The measured duration (μ s).
- Info2: The expected duration (μ s).
- Msg: “UD@Lvl+, SegID=N, ixSeg=I, LastFreq=F”; where ‘N’ is the Segment ID, I is an index into the segment

description in a Segment Array, and F is the measured frequency.

Note: The above “Msg” example is one of several unexpected duration cases that can occur. The exact syntax will vary.

2.1.2.8 Amplitude Fluctuation Event

The Amplitude Fluctuation event (i.e. amplitude modulation) is reported if while in Referenced Mode, the system detects unexpected amplitude changes over a given interval. The test tones in Frontline’s Reference Audio files have a fixed amplitude level over their duration. Therefore, if the corresponding audio levels received over the air by the system fluctuates more than a specified level (this level is based on the received audio stream parameters), then the system generates an Amplitude Fluctuations event.

The system calculates amplitude fluctuations as:

$$\frac{(\text{Max Level} - \text{Min Level})}{(\text{Max Level} + \text{Min Level})} \times 100.$$

The information reported with the Amplitude Fluctuations event is listed below:

- Info1: Percent Amplitude Modulation, expressed as an integer; e.g., “40” = 40%.
- Info2: Not used.
- Msg: Not used.

2.1.2.9 Unexpected Phase Change Event

The Unexpected Phase Change event provides a fine-grained indication of lost or repeated energy. In Referenced Mode, the system knows the Reference Audio file that is being played on the Source DUT; therefore, the system knows when a specific tone should be expected. During this interval, the system checks that the measured average frequency is the same as the expected frequency. If this is correct, the system will continue to monitor the instantaneous frequency. If the instantaneous frequency deviates sufficiently from the current average frequency, the frequency measurement state machine will reset and begin re-measuring.

Typically, the outcome is the discovery of the next scripted (expected) frequency (see [Unexpected Frequency Event on page 10](#) and [Frequency Detection on page 20](#)). However, another outcome can be that the same frequency as the previous average frequency is rediscovered, and this is reported as an Unexpected Phase Change event. Such phase changes are an indicator of losses of signal that do not result in amplitude dropouts, or signal substitution (repetition) of previous audio energy due to things such as “packet loss concealment” tactics.

The information reported with the Unexpected Phase Change event is listed below:

Info1: The phase jump expressed in degrees.

Info2: Not used

Msg: Not used

2.1.2.10 Clipping Event

The purpose of the Clipping event is to report the detection of suspected distortion that occurs when the amplitude of a signal exceeds a digital systems ability to represent it accurately. Clipping is a type of amplitude distortion.

The system reports a Clipping event when consecutive samples at the maximum value that can be represented by the digital system have been detected. Note that the maximum value that can be represented is different depending on the number of bits per sample (i.e. bits of resolution) of the audio stream. The system limits the number of reported Clipping events to typically 10 to 20 per sec.

The number of consecutive samples needed to qualify as a clipping event depends on both sample rate and number of bits per sample. Table 2.2 specifies the number of consecutive samples at the maximum value level that will generate a Clipping event.

Table 2.2 - Clipping Event Threshold

Consecutive Samples	Sample Rate, samples/sec	Resolution, bits
3	8000	16
5	16000	16
11	41000	16
2	64000	16
12	48000	16
24	96000	16

The info reported with the Clipping event is listed below:

- Info1: Contains an integer value representing the RMS average audio level when the clipping event was detected. The value can be converted to a decibel level via the relationship $20 * \text{LOG}_{10} (\text{info1} / 32767)$.
- Info2: Not used.
- Msg: Not used.

2.1.2.11 Excess Noise Event

In Referenced Mode, the system knows the Reference Audio file that is being played on the Source DUT; therefore, the system knows when to expect a period of silence and can measure the noise level during this period. The Excess Noise event is reported when energy sufficiently above the “Silence Threshold” is detected during programmed segments of silence. Excess noise can indicate a poor analog audio chain with an inherently poor noise floor, glitches occurring during silence intervals, or codecs that do not transition to silence instantaneously.

The info reported with the Excess Noise event is listed below. Both info1 and info2 values are integers indicating sample level and can be converted to dBFS via the relationship

$$\text{dBFS} = 20\text{Log}_{10} \left(\frac{\text{info}}{32767} \right)$$

- Info1: The peak noise amplitude (dB).
- Info2: The RMS Average of the noise level (dB).
- Msg: Unused

2.1.2.12 CVSD HF Level Too High Event

The CVSD High Frequency Level Too High event is reported when a CVSD encoded audio stream is detected and there is high frequency energy above 4 kHz that is greater than -20 dBFS.

The information reported with the CVSD High Frequency Level Too High event is listed below:

- Info1: The measured sample level. It can be converted to dBFS via the relationship

$$\text{dBFS} = 20\text{Log}_{10} \left(\frac{\text{info1}}{32767} \right)$$

- Info2: Not used.
- Msg: Not used.

2.1.2.13 End of Test Event

The End of Test event is reported to indicate that the system has completed processing a test script for a Reference Audio file, and that the system has exited Reference Mode. This event is generated when the elapsed time from the start of test is equal to or greater than the scripted duration of a test. It is reached when the number of samples processed equals the number of samples associated with the test duration.

The information reported with the End of Test event is listed below:

- Info1: Unused
- Info2: Unused
- Msg: Unused

2.1.3 Audio Expert System™ Event Type

The following tables list the Audio Expert System™ *Bluetooth*, *Codec*, and audio events with description. Included in the tables is the event severity that can have three values: Information, Warning, and Error. The event severity will appear as icons and text in the Audio Event System once an audio streams has been captured. Refer to [3.2.3 Event Table](#), [Event Table Columns on page 45](#) for an explanation of the severity types.

2.1.3.1 Event Type: *Bluetooth* Protocol

Table 2.3 - Event Type: *Bluetooth* Protocol

Protocol	Severity	Description
A2DP	Warning	AVDTP signal response received for unknown command.
A2DP	Warning	Unrecognized capability type
A2DP	Error	eSCO parameters requested.
A2DP	Error	Profile TX PDUs larger than available bandwidth for active A2DP Streaming interval.
A2DP	Error	Bitpool value does not match configured bitpool range.
A2DP	Error	Attempt to suspend inactive stream.
A2DP	Error	Configuration attempt using unsupported CODEC.
A2DP	Error	Incorrect AVTDP command length.
A2DP	Error	Unknown command Stream End Point Identifier (SEID).
A2DP	Error	A2DP stream configuration attempt using invalid CODEC parameters.
A2DP	Error	A2DP stream configuration request sent during active stream.

Table 2.3 - Event Type: Bluetooth Protocol(continued)

Protocol	Severity	Description
A2DP	Error	Audio data length does not match length header.
A2DP	Error	Incorrect A2DP SBC frame fragmentation.
A2DP	Error	A2DP SBC frame header contents does not match stream configuration.
A2DP	Error	Attempt to configure A2DP stream with unsupported configuration.
A2DP	Error	Reported A2DP stream capabilities do not contain mandatory features.
A2DP	Error	A2DP streaming L2CAP channel not disconnected after ABORT operation.
A2DP	Error	Fragmented AVDTP packet not terminated before sending next packet.
A2DP	Error	Invalid AVDTP transaction ID.
A2DP	Error	Missing AVDTP command response.
A2DP	Error	Unrecognized A2DP content protection type.
A2DP	Error	Attempt to configure delay reporting during incorrect stream state.
A2DP	Error	Attempt to open A2DP stream that has not been configured.
A2DP	Error	Attempt to close A2DP stream that is not active.
A2DP	Error	A2DP streaming channel created before configuration completed.
A2DP	Error	Configuration command contains invalid length parameter.
A2DP	Error	Configuration command contains invalid media transport format.
A2DP	Error	SBC CRC Error.
A2DP	Error	SBC invalid channel mode.
A2DP	Error	SBC invalid header.
A2DP	Error	Invalid AVDTP configuration parameter.
A2DP	Error	Invalid AVDTP stream state

2.1.3.2 Event Type: Codec

Table 2.4 - Event Type: Codec

Codec	Severity	Event	Description
SBC	Information	Codec Initialization	Codec session started
SBC	Information	Codec tear-down	Codec session ended
SBC	Information	Stream Re-configuration	Stream Re-configuration
SBC	Error	Incorrect Configuration Detected	SBC Codec detected a change in audio parameters
SBC	Error	Lost Sync	SBC Codec expected to find synch word: 0x9C instead found: 0x: typically due to corrupted data

Table 2.4 - Event Type: Codec(continued)

Codec	Severity	Event	Description
SBC	Error	Bad Header	SBC Codec detected corrupted header: typically due to corrupted data
SBC	Error	CRC Failure	SBC Codec detected bad CRC: typically due to corrupted data
SBC	Error	No output	SBC Codec generated no output due to corrupted data
mSBC	Information	Codec tear-down	Codec Session Ended
mSBC	Information	Stream Re-configuration	Stream Re-configuration
mSBC	Warning	Packet Loss Concealment	mSBC Codec detected a bad frame and generated substitute data to compensate for it
mSBC	Error	Incorrect Configuration Detected	mSBC Codec detected a change in audio parameters
mSBC	Error	Lost Sync	mSBC Codec expected to find synch word: 0xAD instead found: 0x: typically due to corrupted data
mSBC	Error	Bad Header	mSBC Codec detected corrupted header: typically due to corrupted data
mSBC	Error	CRC Failure	mSBC Codec detected bad CRC: typically due to corrupted data
mSBC	Error	No output	mSBC Codec generated no output due to corrupted data when PLC not configured
AAC	Information	Codec initialization	Codec session started
AAC	Information	Codec tear-down	Codec session ended
AAC	Information	Bitstream type set	The bitstream type has been set. For Bluetooth, it should be LATM.
AAC	Warning	Single frame error, concealment triggered.	During decoding, a single frame error was detected which triggered built in concealment processing.
AAC	Error	Codec setting change	The codec has been re-initialized due to a setting change.
AAC	Error	Unframed stream error	A frame error was detected for an unframed stream. The codec is being reset in order to continue processing.
AAC	Error	Transport not initialized	The codec cannot be initialized for the given transport.

Table 2.4 - Event Type: Codec(continued)

Codec	Severity	Event	Description
AAC	Error	Transport not supported	The selected transport is not supported. This could occur when an out of band LATM is selected opposed to in band.
AAC	Error	Transport failure	General failure in the transport.
AAC	Error	Transport error	This typically occurs when there isn't any configuration information available.
AptX	Information	Codec initialization	Codec session started
AptX	Information	Codec tear-down	Codec session ended
AptX	Error	Bad Data	Non-stereo data has been detected for incoming data stream.

2.1.3.3 Event Type: Audio

Table 2.5 - Event Type: Audio

Test Mode	Severity	Event	Description
Non-Referenced	Warning	Low Volume Alarm	See 2.1.1.1 Low Volume Alarm Event on page 4 .
Non-Referenced	Warning	Clipping	See 2.1.1.2 Clipping Event on page 4
Non-Referenced	Warning	High Volume Alarm	See 2.1.1.3 High Volume Alarm Event on page 5
Non-Referenced	Warning	Dropput	See 2.1.1.4 Dropout Event on page 6
Non-Referenced	Warning	Glitch	See 2.1.1.5 Glitch Event on page 7
Referenced	Info	TestID Found	See 2.1.2.1 Test ID Found Event on page 8
Referenced	Warning	Test Script Not Found	See 2.1.2.1 Test ID Found Event on page 8
Referenced	Error	Invalid Test Script	See Invalid Test Script Event on page 9
Referenced	Error	Synchronization Lost	See 2.1.2.4 Synchronization Lost Event on page 9
Referenced	Error	Unexpected Frequency	See 2.1.2.5 Unexpected Frequency Event on page 10
Referenced	Error	Unexpected Level	See 2.1.2.6 Unexpected Level Event on page 10
Referenced	Error	Unexpected Duration	See 2.1.2.7 Unexpected Duration Event on page 10
Referenced	Error	Amplitude Fluctuations	See 2.1.2.8 Amplitude Fluctuation Event on page 11

Table 2.5 - Event Type: Audio(continued)

Test Mode	Severity	Event	Description
Referenced	Error	Unexpected Phase Change	See 2.1.2.9 Unexpected Phase Change Event on page 11
Referenced	Error	Excess Noise	See 2.1.2.11 Excess Noise Event on page 12
Referenced	Error	Clipping	See 2.1.2.10 Clipping Event on page 11
Referenced	Error	CVSD HF Level Too High	See 2.1.2.12 CVSD HF Level Too High Event on page 12
Referenced	Info	End of Test Event	See 2.1.2.13 End of Test Event on page 13

2.1.4 Supported Codec Parameters

Supported Parameters for SBC Codec

- Sampling Frequencies: 16 KHz*, 32 KHz*, 44.1 KHz, 48 KHz
- Channel Modes: Mono, Dual Channel, Stereo, Joint Stereo
- Block Length: 4, 8, 12, 16
- Number of subbands: 4, 8
- Allocation Method: SNR, Loudness
- Minimum Bitpool Value: 2
- Maximum Bitpool Value: 53

Supported Parameters for MPEG-2, 4 AAC

- Object Types; MPEG-4 AAC LC
- Sampling Frequencies: 44.1 KHz, 48 KHz, 8 KHz*, 11.025 KHz*, 12 KHz*, 16 KHz*, 22.050 KHz*, 24 KHz*, 32 KHz*, 64 KHz*, 88.2 KHz*, 96 KHz*
- Channels: 1 and 2
- Variable Bit Rate and Specified Bit rate

* Audio Analysis not supported . Although, user will be able to play back the audio live.

Supported Parameters for aptX

- Object Types; aptX-classic, aptX-LL (both content protected and non-content protected)
- Audio Format: 16-bit, 44.1kHz
- Data Rates: 352 kbps

Supported Parameters for CVSD

- Channel Mode: Mono
- Sampling Rate: 64 kHz

Supported Parameters for mSBC codec

- Channel Mode: Mono
- Sampling Rate: 16 kHz
- Allocation method: Loudness
- Subbands: 8
- Block Length: 15
- Bitpool: 26

2.2 Referenced Mode Processing

In Referenced Mode, the system operates in a “pseudo closed loop” test scenario where the user plays a specific Reference Audio file (pre-recorded audio test file provided by Frontline) on the Source DUT. The Source DUT negotiates with the Sink DUT to determine the appropriate codec and audio parameters to use and will then process the Reference Audio file accordingly before transmitting the resulting audio via *Bluetooth*.

The Sink DUT receives the encoded audio, decodes it, and processes it for playback. In parallel, Frontline’s system snoops the over-the-air signal between the Source DUT and Sink DUT and emulates the RF reception and decoding done inside the Sink DUT. Frontline’s Audio Expert System automatically detects that a Reference Audio file is being received and then analyzes the resulting audio for deviations from expected parameters.

The Reference Audio files are specific audio files that exercise the system so that audio impairments can more efficiently and accurately be identified and reported. The Reference Audio files are composed of a series of back-to-back and relatively short duration tones of changing amplitude, frequency, and duration.

The analysis of the received audio results in a series of “Audio Events” being reported by comparing changes in the received audio to expected changes of the Reference Audio, and reporting deviation events when they occur.

2.2.1 Transition to Referenced Mode

The system starts up in Non-Referenced Mode, and is continuously looking for a valid Reference Audio file by measuring frequency and amplitude of the received over-the-air audio. Transitioning to Referenced Mode requires the successful detection of a TestID tone sequence of proper frequencies, and durations.

Once the Referenced Mode state is achieved, the expectation is that all tones encountered will conform to the script identified by the collected Test ID digits. The system remains in the Referenced Mode state until either the end of test is reached, or a loss of synchronization occurs (see [Synchronization Lost Event on page 9](#) and [Synchronization on page 19](#)).

2.2.2 Test ID

All Reference Audio files begin with a unique “Test ID” and Frontline’s Audio Expert System is continuously scanning the received audio in search of a valid Test ID. On recognition of a valid Test ID, the “Referenced Mode” state is declared. This state then enables the system to perform more thorough testing of the audio data than can be performed in the non-referenced mode. The system uses the Test ID to determine which test script to use to evaluate the received audio.

A “Test ID” is three digits minimum in length, representing a dot notation “N.vv” Test Identifier. The value ‘N’ may be any length ≥ 1 , indicating a specific test, and “vv” represents a two digit version. Each digit is represented by a tone between 200 and 290 Hz, and is followed by either a 1 kHz delimiter tone or a 400 Hz Test ID terminator. The digit ‘0’ is represented by 200 Hz, the digit ‘1’ by 210 Hz, and so on, up to the digit ‘9’ represented by 290 Hz.

The frequency 400 Hz received following the last digit terminates the digit string. There is no encoding of the ‘.’ (dot character) but test script naming conventions include the string in “N.vv” format, as in “Test_1.01_44.1kHz_16Bit.xml”.

2.2.3 Reference Audio File Structure

The Reference Audio files typically have a structure such as listed in the example below. Such a file structure provides known variations in amplitude and frequency that enable a variety of measurements including the detection of changes to amplitude, frequency, and duration (lost audio, substituted audio).

- Test ID segment
- a 300 Hz tone segment at -12 dBFS for 100 ms
- a 600 Hz tone segment at 0 dBFS (full amplitude) for 50 ms
- a dual tone segment consisting of equal amplitude 880 and 240 Hz tones at -6 dBFS below maximum for 200 ms
- a silence segment for 100 ms
- 600 Hz tone segment which fades out over 100 ms

2.2.4 Synchronization

In Referenced Mode, the synchronization of the received audio (from the Reference Audio files) versus the internal test script is achieved based on changes in frequency of the tones in the Reference Audio file. Frequency changes are used because this parameter is relatively immune to the configuration and typical impairments of the network.

By using changes in frequency, the system is able to continuously re-established synchronization with the Reference Audio file at every tone segment. That is, as each “next” frequency in the script is encountered, the exact time position in the script is re-established, the expectation of the duration of the new frequency is known, and any significant variation before the next frequency change occurs will be reported as an “Unexpected Duration” event.

2.2.5 Minimum and Optimal Amplitude Level Adjustment

When a Test ID Terminator Tone is recognized, its amplitude level is checked to ensure that it is above a minimum “Measurement Threshold” required for accurate operation. To ensure adequate margin, the user must adjust the system under test such that the over-the-air Test ID tone amplitude level is within the acceptable ranges listed below. Note that the system reports the Test ID tone level in the Info1 parameter of the Test ID Found event.

- Speech: -15 dBFS to -3 dBFS
- Music: -20 dBFS to -3 dBFS

In addition to the acceptable amplitude ranges listed above, see [Audio Volume/Level Calibration on page 25](#) and [Adjusting for Optimal Volume Levels on page 28](#) for the optimal over-the-air amplitude levels and description of how properly to adjust the levels .

2.2.6 Amplitude Normalization

Because the exact transmitted (over-the-air) level from the Source DUT devices depends on a number automatic and manual analog and digital gain settings, and because the system needs to determine deviations from expected

levels, Frontline’s Audio Expert System provides a mechanism for automatic level normalization. Normalization is the process of adjusting measured level to account for known gain or loss within the system.

The Test ID Terminator Tone plays a key role in this normalization process. All Test ID tones (in Frontline’s Reference Audio files) are generated at -3 dBFS, a maximum sample value of 23196 for 16-bit resolution. When a Test ID is validated, and the system transitions into Referenced Mode, the measured amplitude of the Test ID Terminator Tone is used to calculate a normalization gain factor. This is the amount of gain required to adjust the measured level of the received audio to -3 dBFS.

This normalization gain is then applied to measured levels before comparing them to expected levels. As an example, if the Test ID Terminator is received at -5 dBFS, 2 dB below the expected value, then the “Normalization Gain” is +2 dB. A subsequent tone scripted at -12 dB, but actually received at -14 dB, would be increased by 2 dB to -12 dB before comparing it to the -12 dB value in the script, and would therefore be accepted as a correct level.

2.3 Basic Measurement Operations

2.3.1 Peak Detection

Detecting peaks is a basic function that underlies all operations because it is common to both amplitude and frequency metrics, and calculations of amplitude and frequency are made continuously by the system.

Sample to sample changes in amplitude are tracked to determine transitions from a positive to negative first derivative (slope). When a peak is detected, its real time level is immediately available for amplitude related measurements. Also, its sample index is saved, and when the next peak is detected, the real time frequency is computed based on the time between peaks.

Frequency, F is the reciprocal of time, t . $F=1/t$, and each sample represents a unit of time equal to $1/\text{Sample Frequency}$.

2.3.2 Frequency Detection

The instantaneous frequency is computed from the received audio based on the discovery of each peak. This is done by maintaining a “last peak” sample number, subtracting that value from the sample number of the current peak, and multiplying by the sample time. For example a 1 kHz tone sampled at 8 kHz precisely at its peaks, will result in 8 samples between each peak. So if a current peak is at sample number 8088, and the previous peak was sampled at sample number 8080, and since the sample time of 8000 Hz is 0.000125 sec, then:

- $(8088 - 8080) * 0.000125 = 0.001$, and
- $1 / 0.001 = 1000$ Hz.

The system maintains a “Frequency Found” state and “Last Average Frequency” variable to aid in the determination of frequency changes. To improve accuracy, the system averages frequency measurements over several samples. This becomes especially important at higher frequencies where there are fewer samples per cycle, and the detected peak may actually be +/- 1 or more samples different than the location of the actual peak.

On each peak after initialization or reset, the measured peak-to-peak frequency is added to an accumulator, and when the configured number of measurements is reached, the average is computed. If the percent change from the “Last Average Frequency” exceeds an audio stream specific limit, an internal frequency change event is declared, and the Frequency Found state is set to true.

On all subsequent peaks, the instantaneous peak-to-peak frequency continues to be computed. If the instantaneous frequency differs from the current average frequency by a sufficient amount, the Frequency Found state is set to false, and the algorithm returns to its initial conditions searching for a stable frequency.

Such “resets” will ultimately result in a reported event of some type. If a new frequency is established, an internal frequency change event occurs. If the same frequency is found as the previous frequency, an “Unexpected Phase Change” event occurs. If the measurement is not successful, a “Synchronization Lost” event will ultimately occur.

2.3.3 Amplitude Levels

Amplitude level measurements include peak and average RMS sample values encountered over the measurement interval. For some of the lower bit rate and bits of resolution codecs, peaks are averaged and reported on a change of the average. For the highest quality settings, a change may be reported on the first occurrence. The measurement intervals are typically 0.05 to 0.1 sec representing 10 to 20 reports per sec. The peak values are commonly used to drive user interface volume level indicators.

The minimum change that will trigger events is configured according to the audio stream specific parameters in effect, but it is generally between 2 to 4 dB.

2.3.4 Crest Factor

The difference between peak and average levels is commonly referred to as the “Crest Factor”. There are some well-characterized relationships between peak and average levels for specific types of audio, and variations from the norm can reveal more subtle aspects of how the audio was processed or what it was subjected to.

When the system is in Referenced Mode, it is receiving audio from which the source is one of Frontline’s Reference Audio files, and the majority of tones in these files are single sine waves. For a sine wave, the RMS level is 0.707 times the peak level, corresponding to a 3 dB difference. Therefore, any difference greater than 3 dB can indicate an abnormal increase in amplitude such as the presences of spikes or other noise, or the presence of multiple frequencies.

Note: Some test segments are intentionally dual tone segments. While Crest Factor for a single sine wave is 3 dB, for 2 sine waves of equal amplitude it is 6 dB.

When the system is in Non-Referenced Mode, there is no pre-defined relationship between peak and average levels. However, there are well-characterized relationships established for speech and audio as listed below. These numbers represent “typical” numbers, and absolute numbers depend on the original audio content.

- Speech: A typical Crest Factor is 12 dB (assuming a signal captured with no dynamic range processing algorithms such as compression or limiting).
- Music: A typical Crest Factor is 18 to 20 dB.

The significance of clipping events reported for arbitrary audio in Non-Referenced mode can be determined by assessing the Crest Factor of the original audio, and verifying the volume settings at the source to ensure sufficient headroom is provided to accommodate that ratio.

2.4 Audio Analysis Concepts

2.4.1 The Decibel

It is useful to understand the use of the decibel. It is simply a convenient way of expressing differences in signal levels. In terms of power, if a signal is reduced to half power we say “it is -3 dB”. If a signal is reduced to half voltage, we say “it is -6 dB”.

A decibel is also convenient for expressing ratios, such as signal to noise ratio. A 20 dB signal to noise ratio sounds like listening to music on a car radio speaker with the windows open and a jack hammer operating close by, or that a 90 dB ratio allows us to listen to even the quietest passages of a symphony with little disturbance.

2.4.2 Units of Measure

Digital representations of analog audio, often referred to as “PCM Data” (Pulse Code Modulation) represent a series of value “snapshots” or “samples” of the analog signal’s instantaneous amplitude (equivalent to voltage) over time. These samples are recorded at a rate of speed referred to as the “Sample Rate”. For example, 8000 samples/sec is typically the minimum for voice material, and 44,100 samples/sec is typically the minimum for music.

Each sample thus represents a signal’s instantaneous amplitude at a point in time, in units equivalent to voltage.

2.4.3 Precision and Dynamic Range

The precision of digitally recorded values depends largely on the resolution of the conversion device, known as an Analog to Digital (A/D) Converter. Typical A/D converters produce samples of 8, 16, or 24-bit resolution.

Table 2.6 - Bit Dynamic Range

Resolution, bits	Max Positive Value	Max Negative Value
8	+127	-128
16	+32,767	-32,768
24	+8,388,607	-8,388,608

The sample values are integers with maximum and minimum values that depend on the number of bits per sample, as related in Table 2.6

One of the characteristics of audio that is most affected by the number of bits per sample is *dynamic range*. It is a term that describes the range of loud to soft audio levels that can be accommodated. The loudest (“full volume”) audio levels are represented by the maximum value that can be represented for a given number of bits per sample. These maximum values equate to the voltage levels of analog amplifiers at which the +/- output voltages reach the limits of the power supplies being used. If a +/- 30 volt power supply is being used, and ignoring output device losses, the output voltage can never exceed +/- 30 volts. If the input signal is increased beyond this, the waveform that would have been above the maximum limit is said to have been “clipped”. A clipped sine wave begins to look like a square wave at extremely over driven levels.

These maximum values are also the values associated with the term “0 dBFS”, which means “0 dB Full Scale”. It is an arbitrary “reference point” that simply means the maximum possible volume that can be represented for a given number of bits per sample (resolution). Therefore, the number of bits per sample determines the maximum values that can be represented, and the maximum values represent the maximum volume.

The lowest level that can be represented is the minimum possible non-zero number that can be represented, and that of course in all cases is the value 1, or -1.

The difference between the maximum and minimum representable values, expressed in decibels is defined as the “dynamic range”.

For any given sample value, Level in dBFS = 20 * LOG (level / max level). For example, for 8-bit samples the minimum level in decibels is:

$$20\text{Log}_{10} \left(\frac{1}{127} \right) = -42 \text{ dBFS}$$

Table 2.7 - Bit Dynamic Range in dBFS

Resolution, bits	Max Value	Dynamic Range
8	127	42 dB
16	32,767	90 dB
24	8,388,607	138 dB

Therefore, for 8-bit samples, the dynamic range is 42 dB, and Table 2.7 indicates the dynamic range for all number of bits per sample (resolution) discussed.

To help give these numbers some meaning, the human ear has a dynamic range of about 120 dB, and since Compact Disks (CDs) use 16-bit samples, they inherently have a dynamic range of 90 dB. The legacy telephone network uses 8-bit samples, and thus has a dynamic range of 42 dB

2.4.4 Measurement Accuracy

The dynamic range of the audio path is important to understand because it has a direct impact on measurement accuracy. It is perhaps easiest to grasp the effect of level if we consider accuracy at the extreme loudest and lowest volumes.

Consider first the accuracy at maximum levels. Looking at the difference in dB between samples of value N and N-1 for several different values of N provides some insight. For 8-bit resolution, the value of N=127 is the max possible value and thus

$$20\text{Log}_{10} \left(\frac{127}{127} \right) = 0 \text{ dBFS.}$$

However, for the value N-1,

$$20\text{Log}_{10} \left(\frac{126}{127} \right) = -0.07 \text{ dBFS.}$$

Therefore, at maximum volume, the difference between N and N-1 is (0 - -0.07), which means we can resolve to within 0.07 dB; This is fairly good.

At half volume, N = 63 and the level in dBFS

$$20\text{Log}_{10} \left(\frac{63}{127} \right) = -6.09 \text{ dBFS.}$$

For N-1 = 62,

$$20\text{Log}_{10} \left(\frac{62}{127} \right) = -6.29 \text{ dBFS.}$$

The difference in this case between N and N-1 is (-6.09 - -6.29) = 0.14 dB.

This is still not so bad, but it becomes apparent that at half volume we can resolve to ½ the accuracy as at full volume, and it follows that at ¼ volume the accuracy will be ¼ that of full volume and so on.

For completeness, consider now taking it to the lower limit by looking at the difference in dB for N and N-1 at their lowest possible non-zero values. These are of course N=2 and N-1=1 and from experience the factor of two is apparent, which translates to a difference of 6 dB. [Just for reference, the difference in dB for the values N=3 and N-1=2 is 3.5 dB (for 8-bit resolution).

So for 8-bit, when a low level analog value close to 2 is sampled it will be quantized to an integer value of 1, 2, or 3 depending on round off or truncation logic in the A/D converter and subject to any noise in the analog signal. The point is that a sample value of 2 in dB can only be stated as “-36 dBFS accurate to within +3.5 or -6 dB”.

In other words, because the accuracy is at best within 3.5 dB and at worst within -6 dB, any measurement that seeks to detect changes accurately of say 3 dB is not possible.

Note: Especially for 8-bit measurements, it is essential that the signal level be as high as possible without exceeding maximum supported values.

2.5 Audio Chain/Network Technologies

Audio data sourced at one device, referred to as “DUT1”, can pass through a variety of audio processing components and transmission channels before it is finally delivered to headphones or speakers at a playback device, referred to as “DUT2”. Figure 2.5 shows an example where a “land line” phone connects to the Public Switched Telephone Network (PSTN) either directly via copper or often through a Private Branch Exchange (PBX). The call is handed off to a Public Land Mobile Network (PLMN), and terminates on a smartphone. In turn, the smartphone establishes a (BT) connection to an in-car speakerphone appliance.



Figure 2.5 - Typical Topology for Voice Scenarios

Figure 2.6 shows another example for music sourced from companies like Pandora and Spotify. A media (audio) server renders the desired audio, which then traverses the public internet, then onto the cellular network of the mobile provider, and is delivered to the smartphone, which sets up the Bluetooth connection to the playback device.



Figure 2.6 - Typical Topology for Music Scenarios

What is important to understand is that the audio can pass through multiple devices and networks. As a result, it can go through multiple conversions from the original analog source, to an initial digital format, and then through multiple additional conversions as it passes through various networks or subsystems. These conversions are referred to as coding and decoding operations, or simply “recoding” operations, and are performed by various codec implementations, which generally perform audio compression.

An analog voice call, for example, may be encoded as μ Law PCM by the Public Switched Telephone Network. Even worse, when IP Telephony is involved the audio will be compressed before it ever leaves the premises. The audio will then be recoded again for transmission through the Public Switched Mobile Network using the provider’s internal Codec technology of choice, and then decoded back to analog by a smartphone. When a Bluetooth appliance is added at the end of the chain, yet another recoding operation is inserted.

For music, similar recoding operations take place as for voice. The original audio is digitally compressed into a format such as MP3, AAC, etc., traverses the internet and PLMN, and is then recoded across the *Bluetooth* channel.

The important point is that *Bluetooth* rendered audio typically passes through multiple coding and recoding operations. The resultant audio is often characterized as having been subjected to the “copy of a copy” syndrome. Also, at various points along the path, digital or analog controls over volume level may exist. Interactions between multiple serial (concatenated) digital processes and improper level settings across the audio “chain” are among the root causes for “poor audio” at the final listening point.

2.6 Audio Volume/Level Calibration

2.6.1 Overview

As mentioned in section [Audio Chain/Network Technologies on page 24](#), the audio that is ultimately delivered to a pair of speakers or headphones passes through a number of processing functions and/or appliances. Some of processing functions include a volume control, and *Bluetooth* playback devices typically have a volume control.

A volume control on the *Bluetooth* source device (DUT1), such as on a smartphone often controls the level of the audio sent over the Bluetooth channel. Indirectly, as a result, it also effectively controls the listening level.

In contrast, a volume control on the *Bluetooth* sink device (DUT2) often controls only the volume level delivered to the speakers. The result is that changing the volume setting may affect only the level delivered to the speakers and has no effect on the level sent over-the air in the *Bluetooth* channel.

As an example, shown in Figure 2.7 is an illustration of a smartphone playing a stored audio file to a *Bluetooth* speaker system. Both phone and speakers have their (vertical slider) volume controls set to maximum. This represents the loudest (but probably distorted) volume the complete system is capable of delivering.

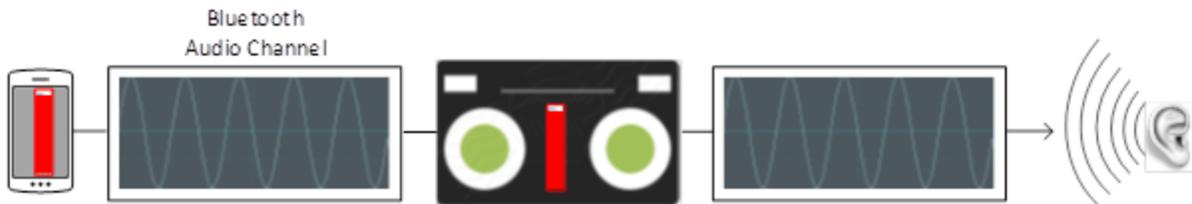


Figure 2.7 - Example 1

Shown in Figure 2.8 is second example with the volume control of the speakers set to half volume, and this results in the level to the listener being reduced to a desirable level. However, note that in this example the volume control of the phone has been left at maximum, thus the over-the-air volume through the *Bluetooth* channel remains at maximum and may be distorted.

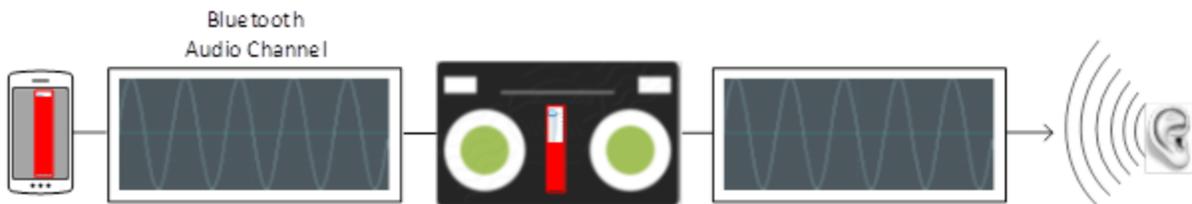


Figure 2.8 - Example 2

Shown in Figure 2.9 is a third example. In this case, in order to reduce the listening volume the slider on the phone has been reduced while the slider on the speakers has been left at maximum. This achieves the same objective as example #2 as far as the listener is concerned. However, reducing the volume at the source (DUT1) also reduces the volume level sent over-the-air through the Bluetooth channel. In effect this reduces the dynamic range that the audio channel is normally capable of accommodating.

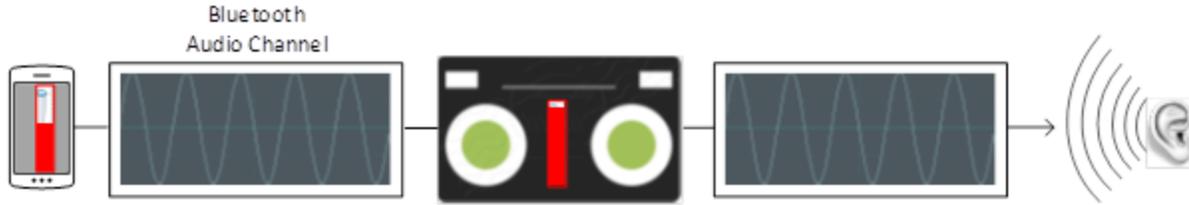


Figure 2.9 - Example 3

Since the Frontline’s Audio Expert System only “sees” the over-the-air audio, it is important that the over-the-air volume be adjusted to a sufficiently high level, but not so high that the signal is distorted. Too much reduction of volume through the over-the-air channel can result in audio levels falling below required Measurement Threshold levels (see [Low Volume Alarm Event on page 4](#) and [High Volume Alarm Event on page 5](#)), and thus making useful audio analysis measurements are not possible.

2.6.2 The General Process

The objective is to achieve settings at the *Bluetooth* Source device (DUT1) that bring the PCM sample levels of tones in the Reference Audio files (for Referenced Mode operation) sent over-the-air as close as possible to the levels at which they were created, without exceeding them. Test ID tones, and the tones in test file sequences for Referenced Mode are generally recorded with a maximum tone segment level of -3 dBFS, although there are a few exceptions where signal levels may be as high as -1 dBFS.

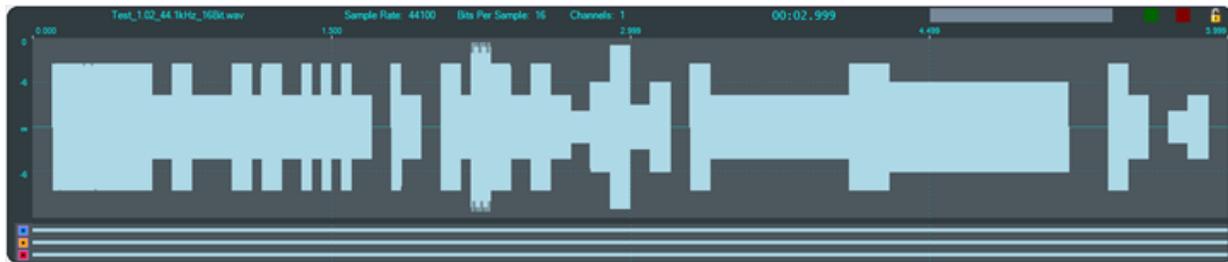


Figure 2.10 - Example: Reference File Waveform, 44.1kHz, 16 bit.

Shown above is an example of the overall envelope of the Reference File. Reference File enables a wide range of tests that includes a number amplitude changes, frequency changes, intentional silence, and multi-frequency tone segments. Its goal is to flush out the audio chain’s general ability to convey amplitude, frequency, silence, and duration

The ideal calibration for this file is one where the waveform visualization on Frontline’s Audio Expert System User Interface (UI) looks identical to the one shown below with respect to maximum levels. In particular, there are three segments in this test whose peaks are at exactly -6 dBFS. That is, there is zero loss or gain through the chain.

Table 2.8 - Example Reference File: -6 dBFS Segments

SegmentID	Frequency, Hz	Start Time, sec.	Duration, sec.
32	800	2.800	0.100
35	1120	3.100	0.100
40	400	4.300	0.900

These -6 dBFS segments are described in Table 2.8. These segments serve as a convenient and quick visual indicator that the levels are appropriate. Especially the longer third case, which is evident at the 4.999 sec reference time in Figure 2.10 (a little over 2/3 of the way through the test).

Show in Figure 2.11 is another graphic of just the first 0.5 sec of the example Reference File, which contains the Test ID value “1.02” (see [on page 8](#) and [Test ID on page 18](#)).

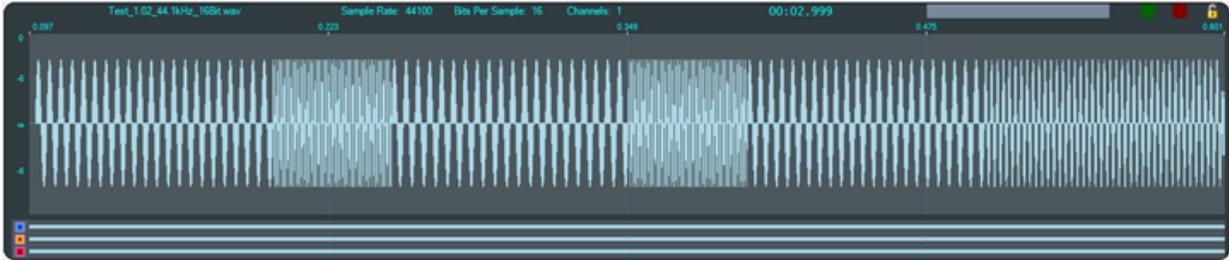


Figure 2.11 - Test 1.02 Test ID Segment

The three digits ‘1’, ‘0’, and ‘2’ are represented by the low frequencies 210 Hz, 200 Hz, and 220 Hz, respectively, which are 100 ms in duration, and are separated by 1 kHz digit delimiters of 50 ms duration. The final tone is a 100 ms segment at 400 Hz, defined as a “Test ID Terminator”. Note that the levels of all of these tones are exactly -3 dBFS.

The peak level of the Test ID Terminator is measured and reported in the “Test ID Found” event’s Info1 parameter (see [2.1.2.1 Test ID Found Event on page 8](#)). It is also serves to calculate the overall gain of the audio channel, up to and including the level of audio on the configured Bluetooth audio channel.

The value in the Info1 parameter of the “Test ID Found” event is optimally the value 23196 and may be converted to dBFS by the relationship:

$$dBFS = 20\text{Log}_{10} \left(\frac{\text{Info1}}{32767} \right) .$$

Optionally the value can be converted to “Channel Gain” via the relationship

$$dB = 20\text{Log}_{10} \left(\frac{\text{Info1}}{23196} \right) .$$

Table 2.9 - "Test ID Found" Event "info1" Maximum and Minimum Values

Format	Application	Maximum	Minimum
Integer	Speech	23196	5826
	Music	23196	3297
Level	Speech	-3 dBFS	-15 dBFS
	Music	-3 dBFS	-20 dBFS
Chanel Gain	Speech	0 dB	-12 dB
	Music	0 dB	-17 dB

Table 2.9 indicates the maximum and minimum acceptable levels for the “Test ID Found” Info1 parameter in integer form, decibel level in dBFS, and Channel Gain in dB.

Example 1: For the case where the Info1 parameter is converted to “Channel Gain”, if the audio is speech (i.e. transported via a SCO channel), then a value of -11.9 dB is acceptable, and a value of -12.1 dB is not.

Example 2: For the case where the Info1 parameter is converted to “Channel Gain”, if the audio is music (i.e. transported via an A2DP connection), then a value of -16.9 dB is acceptable, and a value of -17.1 dB is not.

dB is acceptable, a value of 0.1 dB is not.

For both cases, at the high volume end, a value of -0.1

2.6.3 Adjusting for Optimal Volume Levels

The exact steps that need to be taken depend on the exact devices being used, and their device specific setup requirements, and the speech or audio configuration under test. For the simplest case where, for example, a “music” audio file is to be played by a smartphone to a set of *Bluetooth* speakers, the typical steps would include the following.

1. Choose an audio reference file to be played at DUT1 appropriate for the configuration to be tested.

The test files are stored on the users computer In the directory "`\Frontline <version #>\Development Tools\Audio Expert Test Files\`". For example,

`Test_1.03_48kHz_16Bit_3Loops_2Ch.wav`

Note: Reference test files are periodically updated. Shown here is an example. Files delivered with your latest Frontline software version may have changed. Contact Frontline Technical Support for information on the latest reference file versions.

2. Before establishing the *Bluetooth* connection, play the file while listening to it on the DUT1 device itself, and become familiar with the overall sound quality, generally ignoring exact volume.
3. Set the playback volume at DUT1 to maximum.
4. Set the playback volume at DUT2 to minimum.
5. Establish the *Bluetooth* connection and begin playback of the file on DUT1, if possible in “Loop” or “Repeat” mode to avoid having to continuously restart.
6. Slowly increase the volume on DUT2 until it is at a comfortable level.
7. If the audio sounds distorted, reduce the playback volume at DUT1, and repeat Step 6.
8. When the clarity of the audio is comparable to that heard when listening to the DUT1 device, proceed with using the Frontline software enabled to capture and analyze the Bluetooth data.
9. Visually observe the waveform in the Audio Expert System **Wave Panel** comparing it to the image above, Figure 2.10. If the level of the -6 dB, 0.9 sec duration, 400 Hz tone (a little over 2/3 of the way through the test) is grossly above or below the -6 dB (50% volume) grid line, adjust the DUT1 volume accordingly and repeat this step. Optimally it would be on or just below the -6 dB gridline, but not above. The peak should never hit the maximum positive or negative limits of the display.
10. Find the “Test ID Found” event in the **Event Table** to verify that the system has transitioned to Referenced Mode, and verify that the value for “Channel Gain” (or “Level” as implemented in the UI) is within the range of values specified in Table 2.9.

If the observed (captured) waveforms do not reasonably conform to the above graphic for Test_1.02, or the “Test ID Found” event is not reported, there is a problem along the audio chain. This could be as simple as a configuration setting, or more subtle such as an encoder/decoder incompatibility.

2.7 Audio Rating Metric™

Frontline's Audio Rating Metric™ (ARM) is a process for rating audio impairments or differences resulting from problems in *Bluetooth* audio channels. In the Audio Expert System™ ARM operates only in the reference mode because it produces an objective score based on a set of deterministic measurements of captured reference audio as compared to the original reference audio.

ARM is a real-time process. Individual ARM scores are calculated for each sub-segment of the reference audio. Therefore, the scores are displayed as an overlay on the captured audio waveform providing visual traces of the audio channel quality. For most sub-segments, scores are computed when a change of frequency is detected, and for silent segments, for multi frequency segments, or for level changes. Supported test files can change frequency as often as every 50 milliseconds which equates to ARM's smallest real-time response. However, typical segments resulting from frequency change would occur every 100 milliseconds, as well as a few longer segments.

There are six components to the ARM segment score computations. These components are listed and described in [See Audio Rating Metric Score Components below](#). Each metric score is based on the percent change formula

$$\frac{A-B}{A+B}$$

where A and B are the two component values being compared. This computation always results in values between 0.0 and 1.0.

Each component score is multiplied by an associated weighting factor. The exact weight value for each component depends on its relevance with respect to audible degradation, and the sum of the individual weight values must total exactly 1.0. Having prescribed a set of component weights, all audio measured is subjected to the same judgments, and the score outcome of a given audio sample is repeatable.

Table 2.10 - Audio Rating Metric Score Components

Score Component	Comments
Frequency Resets	<p>This component contributes sensitivity to lost continuity, unexpected changing frequency, or other forms of (abnormal) frequency deviation.</p> <p>As each audio waveform peak occurring above Measurement Threshold is detected, the time between it and the previous peak is used to calculate the instantaneous peak-to-peak (p-p) frequency associated with the interval. These instantaneous frequencies and peak amplitudes are also used to generate running averages of frequency and amplitude. Whenever the instantaneous frequency deviates from the average by more than a configured percentage, a "Reset" is declared, and re-acquisition of average frequency begins. This event occurs naturally in reference audio, which includes numerous transitions between low and high frequencies; but normally a reset occurs only once per scripted transition. In the face of impairments such as lost data, events such as "Phase Change" and/or "Unexpected Duration" may be reported.</p>

Table 2.10 - Audio Rating Metric Score Components (continued)

Score Component	Comments
Frequency Measurement Restarts	<p>This component is an indicator of significant frequency instability.</p> <p>Following a Frequency Reset, the Audio Expert System™ begins tracking just the instantaneous p-p frequency, and comparing each measurement to the previous measurement. In all scripted reference audio, frequency transitions are abrupt and occur at 0 degrees. That is, the frequency changes instantaneously. However, various encoding/decoding operations, and in particular combinations of concatenated encoding/decoding operations, can result in a variety of unanticipated interactions. They can result in “cross fading” from one frequency to the next in some cases (a “fade out” of one frequency, and a “fade in” to the next). Also, some codecs employed in mobile networks do not convey instantaneous transitions from one frequency to another. The first cycles of a new frequency may be higher or lower than the transmitted frequency, “settling” to the transmitted frequency after some number of cycles, essentially exhibiting some degree of frequency deviation over the duration of the segment.</p> <p>These cases result in p-p frequency measurements with continuously changing outcomes as a result of the “beating” of multiple frequencies, or frequency deviations.</p> <p>Whenever a p-p change occurs during the reset state which is greater than a configured percentage, a “restart” of the acquisition process occurs, and this score component tracks the percent difference between the number restarts anticipated per segment, and the actual number encountered.</p>
Frequency Difference	<p>This score component contributes the percent difference between measured and scripted frequencies.</p> <p>When each frequency acquisition process concludes, reaching a steady state average frequency, an internal “Frequency Change” event is declared. When this occurs script tracking advances to the next segment of the script, and a comparison of the measured to scripted frequency is performed.</p>
Level Difference	<p>When the Audio Expert System™ detects a significant level change, it compares the new level to the scripted level, and this score component contributes the percent difference between the two.</p>
Amplitude Modulation	<p>This score component contributes the maximum Amplitude Modulation measured over each segment. In effect, it provides a measure of amplitude stability.</p>
Duration Difference	<p>When each new segment is encountered the duration of the previous segment is measured and compared to the scripted duration. This score component contributes the percent difference.</p>

The maximum ARM score is 5.0 and descends to a minimum of 1.0, in a manner similar to the scoring common to most “Mean Opinion Scores” (MOS). “Errors” found by each of the detection mechanisms effectively “deduct” from the maximum. The implications of the score values with respect to hearer’s perceived audio quality are indicated in the following [See Audio Rating Metric Scale Interpretation below](#).

Table 2.11 - Audio Rating Metric Scale Interpretation

Major Scale Point	Quality	Hearer's Perception of Difference
5	Excellent	Imperceptible

Table 2.11 - Audio Rating Metric Scale Interpretation (continued)

Major Scale Point	Quality	Hearer's Perception of Difference
4	Good	Perceptible but not annoying
3	Fair	Slightly annoying
2	Poor	Annoying
1	Bad	Very annoying

The ARM score computation implements a simple linear scaling formula.

$$ARM = ARMfactor \times \sum_{i=1}^6 (Score\ Components)_i + ARMBaseScore$$

Where,

ARMfactor = a scaling factor to achieve an ARM range of 1.0 to 5.0.

ARMBaseScore = 5.0.

The percent change formula naturally produces an exponential-like curve for each of the component scores. The ARMfactor inverts it to look appear as in [See Audio Rating Metric Score Chart below](#).

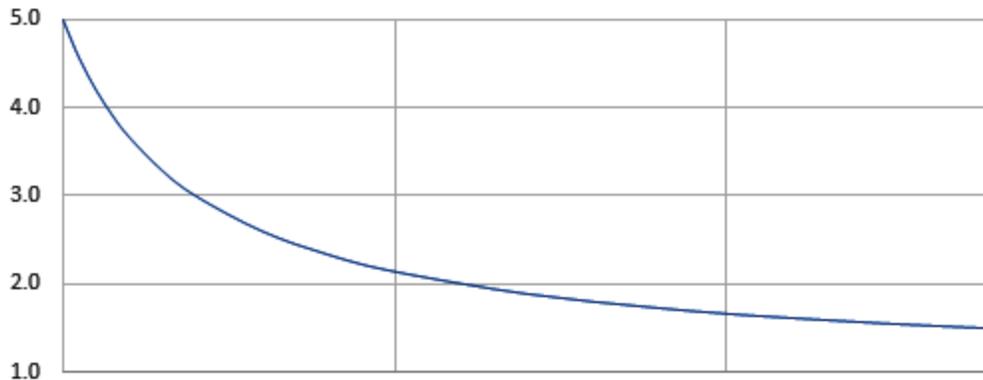


Figure 2.12 - Audio Rating Metric Score Chart

The resulting general behavior is that score reduction tends to follow a somewhat logarithmic behavior. Exactly what it takes for any particular component to reduce score from 4, to 3, to 2, etc., depends on the value being measured. But as one example, the number of Frequency Resets that should occur for perfect audio is exactly 1 per frequency change. But, if there are phases changes or other degradations Frequency Resets may increase. Imagining that frequency resets are the only component in the score, the resulting scores for increasing resets would be as shown in [See Audio Rating Metric Example: Frequency Reset Impact below](#).

Table 2.12 - Audio Rating Metric Example: Frequency Reset Impact

Resets	1	2	3	4	5	6	7	8	9	10
ARM Score	5.0	3.7	3.0	2.6	2.3	2.1	2.0	1.9	1.8	1.7

The effective behavior is that the score is fairly sensitive to any initial deduction (impairment), but becomes less sensitive as deductions (impairments) continue to occur. For example, as indicated above, 2 additional unscripted

frequency resets (for a total of 3) drops the score by 2 full points to 3.0. But 3 times this, or 6 resets for a total of 7, are required to drop the score by another full point to 2.0.

Chapter 3 Audio Expert System™ User Guide

This section provides information on the Audio Expert System™ user interface and controls. This information can also be found in the Frontline software Help. Also this information is contained in the Frontline BPA 600 and Frontline Sodera On-line Help and in the Frontline BPA 600 and Frontline Sodera User Manual PDF document.

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3.1 Starting the AudioExpert System (Sodera and BPA 600 only)

To use the Audio Expert System™, the user must have

- Current Premium Maintenance purchased from Frontline
- Frontline hardware, with Audio Expert System™ license installed, connected to the PC. This is a requirement for both live capture and when viewing a saved capture file.

For live capture, set up the Frontline Sodera or Frontline BPA 600 datasource and begin capturing data.

Note: Proper positioning of the Frontline hardware relative to the devices under test (DUT1-source, DUT2-sink) will contribute to effective data capture. [Air Sniffing: Positioning Devices on page 53](#).

For viewing a capture file, load the saved file from the **Control** window **File** menu.

When an audio stream is available the open the **Audio Expert System™ Window** by clicking on the **Control** window Audio Expert System™ button . If the Frontline hardware is not licensed for Audio Expert System™, the button will not be present.

3.2 Audio Expert System™ Window

(Click here to see an introduction video...)

This window is the working space for the Audio Expert System™. Upon opening Audio Expert System™ the window shown below will open with four main areas displayed :

- Global Toolbar - Provides play cursor controls, waveform viewing controls, and volume controls that affect all Wave Panels.
- Wave Panel - Displays the waveforms for each captured audio stream. There is a separate Wave Panel for each stream. Each panel contains local information, controls, and an event timeline specific to the displayed audio stream being shown. Other Wave Panels that may be off screen may be viewed using the vertical scroll control or by collapsing other Wave Panels.
- Event Timeline - The Event Timeline shows *Bluetooth* events, Codec events, and Audio events synchronized to the displayed waveform. There is an Event Timeline in each Wave Panel.
- Event Table – A tabular listing of *Bluetooth*, codec, and audio events with information on event severity, related *Bluetooth* frame, timestamp, and event information.

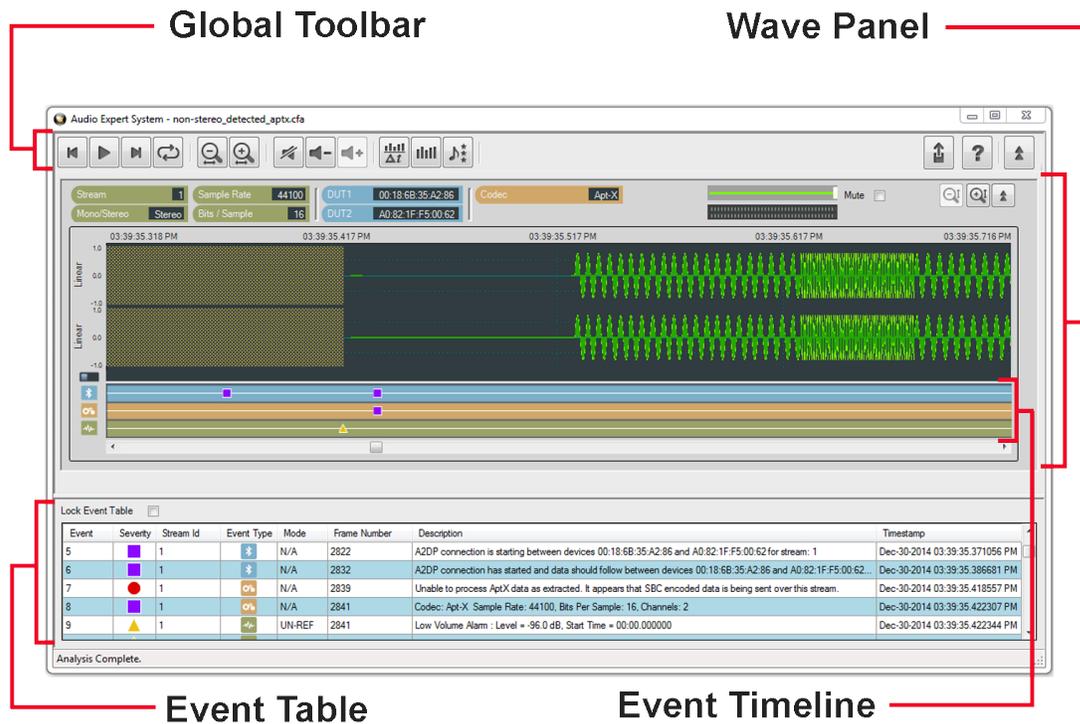


Figure 3.1 - Audio Expert System™ Window

Color Codes and Icons

The Audio Expert System™ uses standard color codes and icons to assist the user in focusing on specific issues.

Table 3.1 - Audio Expert System™ Color Codes and Icons

Category	Sub-Category	Color Code	Icon
Technology	Bluetooth	blue	
	Codec	orange	
	Audio	green	
Event Severity	Information	purple	
	Warning	yellow	
	Error	red	

Note: If an Event Severity icon is surrounded by a dark line, the event is a global event and not applying to a particular captured waveform. The event is assigned to "Stream 0" in the Event Table.

The following topics describe the Global Toolbar, Wave Panel, Event Timeline and Event Table in more detail.

3.2.1 Global Toolbar

[\(Click here to see an introduction video...\)](#)

The global toolbar provides audio play controls, audio play cursor positioning controls, waveform viewing controls, and volume controls. Global toolbar controls apply simultaneously to all waveform panels.



Table 3.2 - Global Toolbar Controls

Icon	Description
	Home: Moves play cursor to beginning of the waveform
	Play : Start playing the audio from the current play cursor position. Toggles to Pause when clicked. Pause: Stops audio play back at its current position, toggles to Play when clicked.
	End: Moves the play cursor to the end of the waveform

Table 3.2 - Global Toolbar Controls (continued)

Icon	Description
	<p>Loop: Loops waveform playback continuously. If the Play button is visible it will toggle to the Pause. Clicking the Pause button will stop Loop playback. Clicking on the Loop button will stop the loop and the playback. If there is a selection on the waveform, only the selection will loop.</p>
	<p>Horizontal Zoom Out: Increases the amount of data that is visible on the screen; however, less detail is discernible.</p>
	<p>Horizontal Zoom In: Decreases the amount of data that is visible on the screen; however, more detail is discernible</p>
	<p>Lock/Unlock (Operational in live mode only): Selecting Lock will freeze the waveform display; however, the Audio Expert System™ will still continue to analysis new audio data.. Selecting Unlock will jump to the waveform end and then resume following the waveform.</p>
	<p>Mute: Mute will mute / unmute audio playback for all Wave Panels. Individual Wave Panel Mute control will override the Global Toolbar Mute for that panel only.</p>
	<p>Volume Down: Decreases the audio playback volume of all Wave Panels based on the current volume level setting for each individual Wave Panel.</p>
	<p>Volume Up: Increases the audio playback volume of all Wave Panels based on the current volume level setting for each individual Wave Panel.</p>
	<p>Average Bit Rate Overlay: Displays an overlay graph of the average bit rate for the audio stream in each Wave Panel. The average is based on a 0.10 second moving window. When active, will deactivate Actual Bit Rate Overlay .</p>
	<p>Actual Bit Rate Overlay: Displays an overlay graph of the instantaneous bit rate for the audio stream in each Wave Panel. When active, will deactivate Average Bit Rate Overlay</p>
	<p>Export Data: Exports audio data in .raw and/or .wav format for selected Wave Panels or all the Wave Panels. This button also lets user export Event Table data in .csv format. Refer to Waveform Export Audio Data for more details .</p>

Table 3.2 - Global Toolbar Controls (continued)

Icon	Description
	Help - Opens Frontline software help.
	Collapse/Expand: Toggles between collapsing and expanding all Wave Panels. Note that the Wave Panel Local Controls Collapse/Expand control will locally override the Global Toolbar Collapse/Expand control.

3.2.2 Wave Panel

[\(Click here to see an introduction video...\)](#)

The Stream Panel is where the details of the captured audio stream are presented. The Stream Panel displays the captured audio waveform along with an event timeline that displays discrete *Bluetooth*, *Codec*, and *Audio* events synchronized to the captured waveform. .

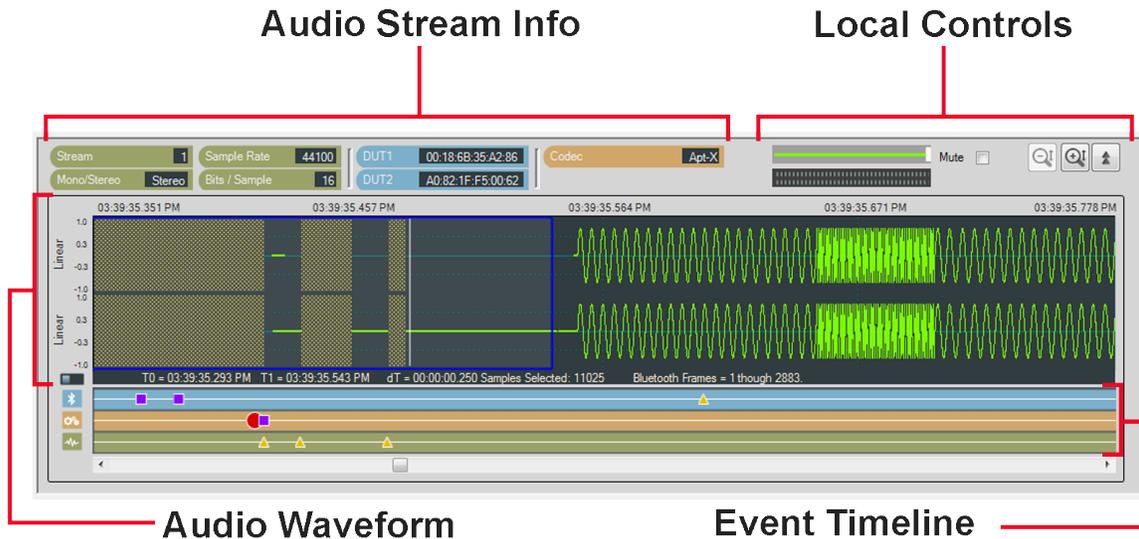


Figure 3.2 - Wave Panel

The Wave Panel contains four sections.

1. Audio Stream Info that provides users with information, such as sample rate, bit/sample, codec and DUT (Device Under Test) addresses.
2. Local Controls include audio volume controls and Indicators, “Mute”, “Vertical Zoom” and “Collapse/Expand”
3. An Audio Waveform which is plotted as amplitude (linear or dB) versus time and an interactive play cursor. The play cursor appears as a white vertical line across the waveform.

- Event Timeline that shows color coded *Bluetooth* , *Codec* , and *Audio*  events. Details of these events are listed in the Audio Expert System™ Event Table.

3.2.2.1 Audio Stream Info

The Audio Stream Info displays Audio, *Bluetooth*, and *Codec* information (left to right in the image below) about the audio waveform displayed in the panel. This information is discovered during AVDTP signaling when the devices under test (DUT) negotiate audio streaming parameters.

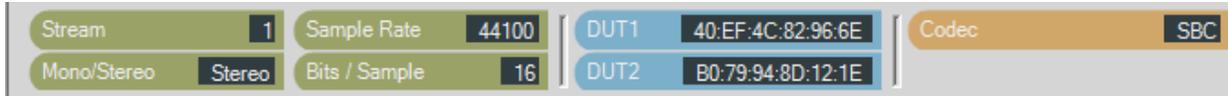


Figure 3.3 - Audio Stream Info in the Wave Panel

Table 3.3 - Audio Stream Info Tags

Category	Name	Description
Audio	Stream	A system assigned index number that represents an audio waveform between a pair of Bluetooth devices. This number appears in the Event Table for easy cross-referencing.
	Sample Rate	Displays the sampling frequency used to digitize the original audio.
	Mono/Stereo	Indicates if the audio data is monaural or stereophonic.
	Bits/Sample	Displays the number of bits per sample of the audio data.
<i>Bluetooth</i>	DUT1	<i>Bluetooth</i> address of one device in the connection. Can be either sending or receiving the audio data.
	DUT2	<i>Bluetooth</i> address of the other device in the connection. Can be either sending or receiving the audio data.
<i>Codec</i>	Codec	Displays the Codec type used by the captured audio stream. The supported codecs include SBC, AAC, aptX, mSBC, and CVSD.

SBC Codec Information Pop-up

When you hover over the **Codec** tag and the Codec = SBC a pop up will appear that shows additional information about which SBC parameters can be used. The pop-up is visible as long as the cursor hovers over the **Codec** tag.

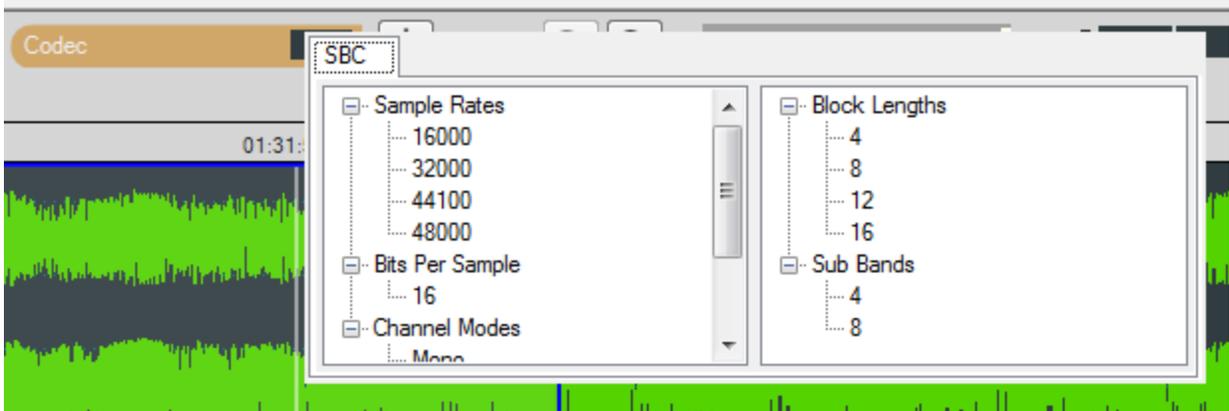


Figure 3.4 - SBC Codec Information Pop-Up on Cursor Hover Over

3.2.2.2 Local Controls

The Local Controls in each Wave Panel provide the user with indicators and controls for waveform display and audio play back.

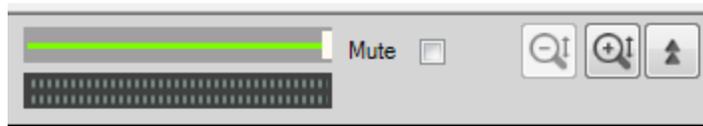


Figure 3.5 - Wave Panel Local Controls

Waveform Play Back Volume



The volume slider controls the playback volume for the audio in each Wave Panel.

Audio Volume Indicator



The volume indicator shows the relative audio volume at the waveform display play cursor. When the green bars completely fill the indicator the audio volume is at its highest level. As the volume decreases, the bars will move to the right linearly, with no visible green bar indicating no audio. The volume indicator will continue to operate if the audio stream has been muted.

Mute



Checking the **Mute** check box will silence the Wave Panel's audio output. The volume indicator will respond to the audio volume but nothing will be heard. All panels can be simultaneously muted using the Audio Expert System™ Global Toolbar. The Wave Panel mute is a local control only. However, the Global Toolbar mute control will set the Stream Panel's Local Controls mute.

Vertical Zoom



Each Wave Panel contains local Vertical Zoom controls that expands or reduces the waveform display vertically. The waveform amplitude is always visible, and the Vertical Zoom controls increases or decreases the entire vertical size of the display. The vertical zoom buttons will turn gray and become inactive when the maximum and minimum values are reached.

Collapse/Expand Control



Collapse/Expand button toggles between two views. The top image indicates that the Wave Panel is expanded. When the bottom image is visible it indicates that the Wave Panel is collapsed.



When the top image is visible, clicking on it will collapse the Wave Panel to the minimum size that shows only the Stream Info and the Local Controls. When the bottom image is visible, clicking on it expands the Wave Panel to full size.

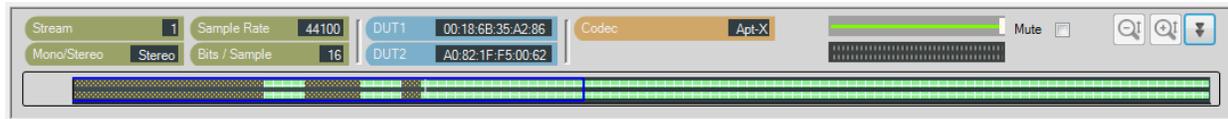


Figure 3.6 - Collapsed Wave Panel

3.2.2.3 Audio Waveform Panel

The Audio Waveform Panel displays the captured audio waveform. If the waveform is stereo, both channels are visible in the Wave Panel. The user can view the entire waveform or can zoom to view a portion of the waveform in more detail.

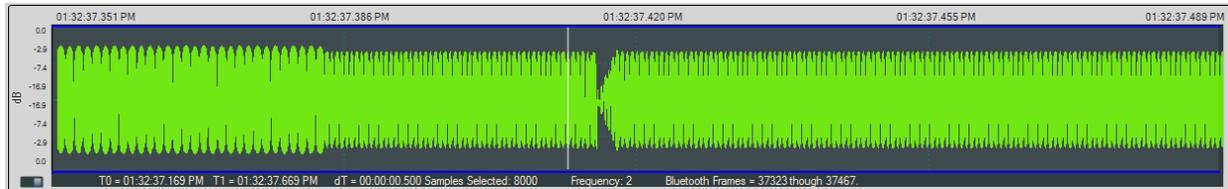
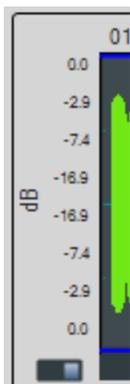


Figure 3.7 - Audio Waveform Panel in the Wave Panel

Table 3.4 - Global Toolbar Waveform Horizontal Zoom Controls

Control	Description
	Horizontal Zoom: Increases the amount of data that is visible on the screen; however, less detail is discernible.
	Horizontal Zoom: Decreases the amount of data that is visible on the screen; however, more detail is discernible.

Waveform



The audio waveform is plotted as amplitude versus time on the Wave Panel. The amplitude scale is located on the left edge of the Wave Panel. The waveform's amplitude can be linear or in decibels. The linear range is -1.0 to +1.0. The range for the dB scale is 0 dB for the maximum positive and maximum negative values, and silence is negative infinity. A toggle switch at the bottom of the amplitude scale will switch between **Linear** scale and **dB** scale. Moving the switch to the left will display the **Linear** scale and moving it to the right will display the **dB** scale.

Play Cursor

The Play Cursor is identified by a white vertical line on the Wave Panel. The Play Cursor appears when user clicks on any point in the waveform, or, if the cursor is already present it can be dragged to another position. To drag the Play Cursor, hover the mouse cursor over the Play Cursor until the mouse cursor changes to a pointing hand; click and drag the cursor to a new position.

Waveform Segment Selection

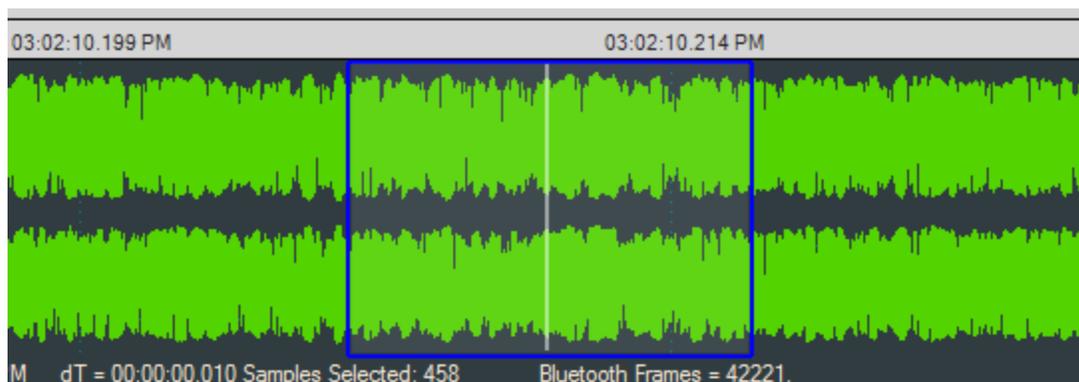


Figure 3.8 - Selection in the Audio Waveform

A waveform segment selection is identified by a blue border surrounding the selection. Procedures for selecting a segment depend on the desired actions.

Table 3.5 - Segment Selection Procedures

Desired Action	Procedure
Loop play back	1. Zoom in to the waveform segment of interest. 2. Click in the approximate center of the proposed selection. This will place the Play Cursor in the area to be selected. 3. Move the mouse cursor to the right or left of the Play Cursor, click and hold, then drag over the waveform segment of interest. Release the mouse key. The selection is surrounded by a blue border.
View waveform details	1. Zoom in to the segment of interest. 2. Move the mouse cursor to the right or left limit of the waveform segment of interest; click and hold, then drag over the waveform segment of interest. Release the mouse key. The selection is surrounded by a blue border.

For either of the procedures described in the table above, once the selection is made details of the segment appear below and to the left of the waveform. These details include selection start and stop range ("T0" and "T1"), the time difference ("dT"), samples selected, frequency, and "Bluetooth Frames" selected.

Right-clicking in the Waveform panel will open a pop up menu (see [Wave Panel & Event Table Pop-up Menu on page 46](#)). Selecting **Zoom to Selection** will expand the selection to the full width of the Wave Panel. Other selection options in the pop up are **Select Area**, **Clear Selection**, and **Copy Selection**.

Actual Bitrate Overlay Display

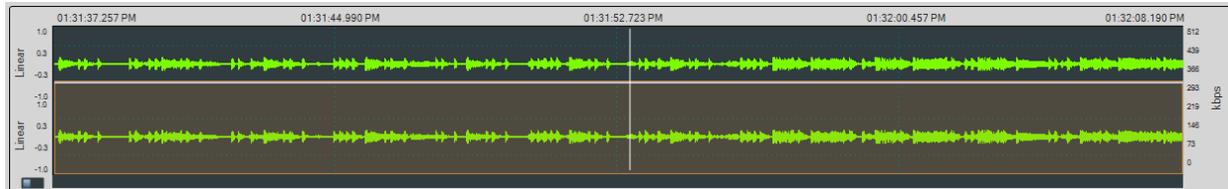


Figure 3.9 - Actual Bitrate Overlay

The Average and Actual audio stream bitrate graphs can be displayed over the audio waveform using the Global Toolbar Average Bitrate Overlay  and Actual Bitrate Overlay  buttons respectively. These are presented as overlays onto the main Wave Panel so the user can correlate audio issues with bitrate changes and the like. The scale is in kbps (kilo bits per second). Hovering over the bitrate scale will display a pop-up showing the bitrate at the play cursor position.

Actual Bitrate is based on the throughput at the Codec level.

The Average Bitrate is the moving average over 0.1 sliding-second window.

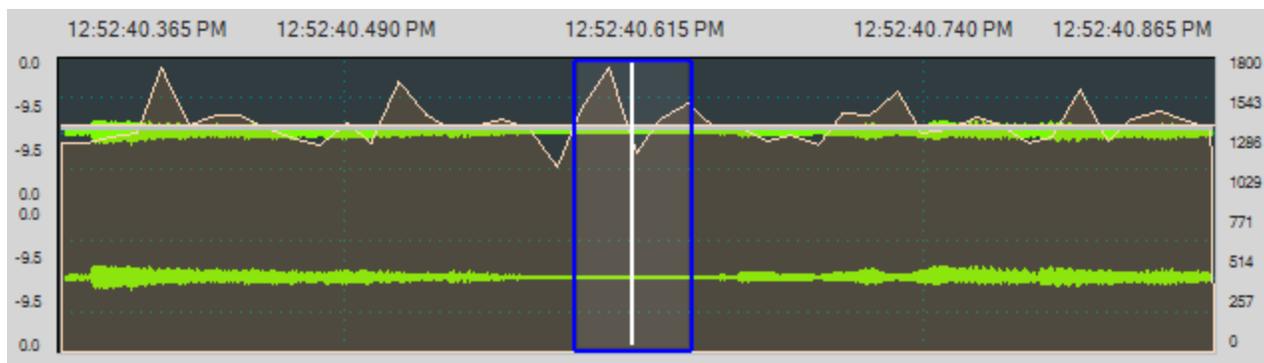


Figure 3.10 - Average Bitrate Overlay

All of the information for calculating the Actual and Average Bitrate is in the codec data frame header.

3.2.2.4 Event Timeline

([Click here to see an introduction video...](#))

The Event Timeline in the Wave Panel shows the *Bluetooth* , *Codec* , and *Audio*  events related to the waveform being viewed. The events are synchronized in time to the waveform displayed in the Wave Panel. The event severity is displayed as Information , Warning , and Error .



Figure 3.11 - Event Timeline Shown with Wave Panel

Clicking on an event in the Event Timeline shows a relevant selection in the Audio Waveform Panel. The size of the selection depends on the number of frames associated with the selected event. This selection will appear in all Wave Panels; however, the event severity icon will only appear in the Wave Panel associated with the event.

To assist the user with viewing events in detail, the Event Timeline will zoom in and out in sync with the Wave Panel.

Event Timeline Example

This example shows that event 159 was selected in the Event Table resulting in the severity icon being enlarged in the Event Timeline. The system automatically selected the surrounding area—the blue outline.

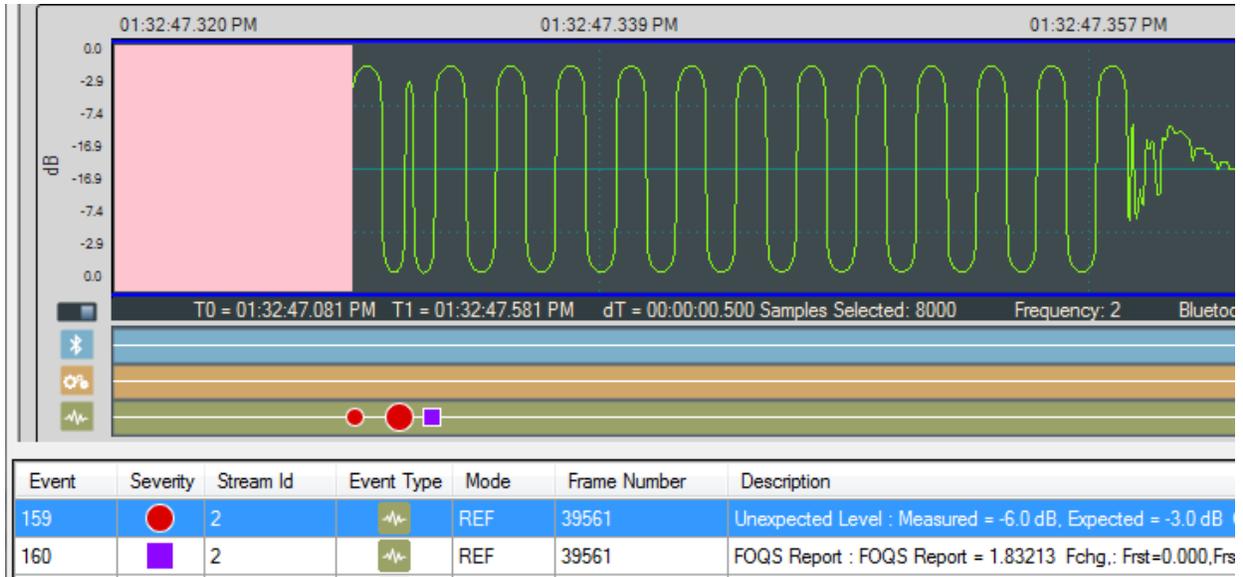


Figure 3.12 - Example: Event Table Selection Shown in Event Timeline

Event Pop Up

When the cursor hovers over a selected event severity icon in the Event Timeline, a pop-up will display the event class, severity, and associated *Bluetooth* frame.

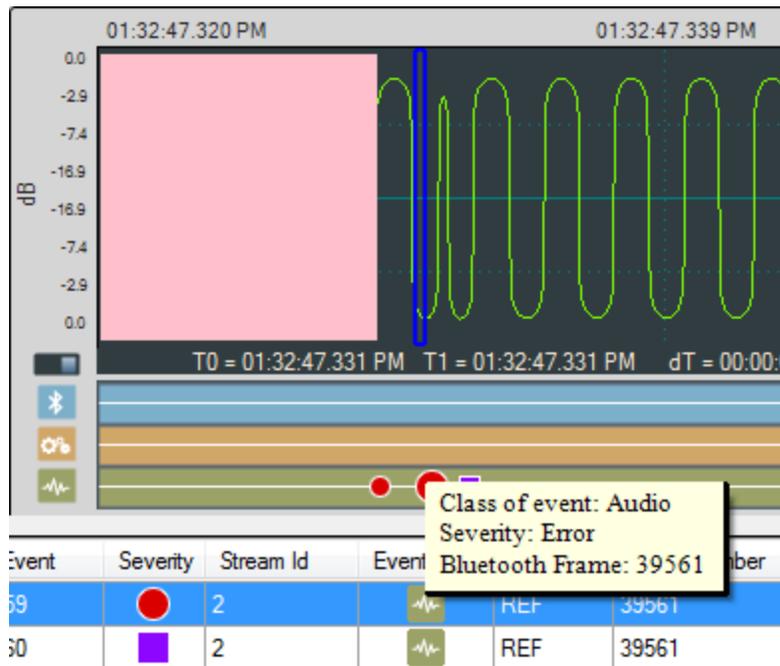


Figure 3.13 - Event Timeline Selected Event Pop Up

3.2.3 Event Table

[\(Click here to see an introduction video...\)](#)

The Event Table lists all audio stream events . Clicking on an event will select that event in the Event Timeline in the Wave Panel. If the selected event is outside the visible area of the waveform, the waveform will move and bring the selected event to the center of the display. The event icon in the Event Timeline is also centered and the selected icon will be larger than the non-selected event icons. Selecting one or more events in the table will highlight the associated frames in the standard Frontkline software windows, such as **Frame Display**, **Coexistence View**, **Bluetooth Timeline**, etc. .

Event	Severity	Stream Id	Event Type	Mode	Frame Number	Description	Timestamp
17	▲	1	🔗	N/A	3039	Packet retransmission.	Mar-31-2014 12:52:38.080991 PM
18	■	1	🔗	N/A	4094	A2DP paused between devices 98:0D:2E:23:B6:2E and 00:07:62:0F:00:00 for stream: 1 using codec: SBC	Mar-31-2014 12:52:45.553569 PM
19	■	1	🔗	N/A	4095	A2DP paused between devices 98:0D:2E:23:B6:2E and 00:07:62:0F:00:00 for stream: 1 using codec: SBC	Mar-31-2014 12:52:45.617944 PM
20	▲	0	🔗	N/A	4101	SCO connection request.	Mar-31-2014 12:52:46.151071 PM
21	■	2	🔗	N/A	4105	SCO connection established between devices 98:0D:2E:23:B6:2E and 00:07:62:0F:00:00 for stream: 2 using cod...	Mar-31-2014 12:52:46.504191 PM
22	■	3	🔗	N/A	4105	SCO connection established between devices 00:07:62:0F:00:00 and 98:0D:2E:23:B6:2E for stream: 3 using cod...	Mar-31-2014 12:52:46.504191 PM
23	■	2	🔗	N/A	4108	Codec: CVSD Frequency: 64000, Bits Per Sample: 16, Channels: 1	Mar-31-2014 12:52:46.806067 PM
24	■	3	🔗	N/A	4256	Codec: CVSD Frequency: 64000, Bits Per Sample: 16, Channels: 1	Mar-31-2014 12:52:47.357946 PM
25	■	2	🔗	N/A	13222	SCO disconnected between devices 98:0D:2E:23:B6:2E and 00:07:62:0F:00:00 for stream: 2 using codec: CVSD	Mar-31-2014 12:53:04.151789 PM
26	■	3	🔗	N/A	13222	SCO disconnected between devices 00:07:62:0F:00:00 and 98:0D:2E:23:B6:2E for stream: 3 using codec: CVSD	Mar-31-2014 12:53:04.151789 PM
27	■	1	🔗	N/A	13253	A2DP resumed between devices 98:0D:2E:23:B6:2E and 00:07:62:0F:00:00 for stream: 1 using codec: SBC	Mar-31-2014 12:53:05.446738 PM
28	■	1	🔗	N/A	13254	A2DP resumed between devices 98:0D:2E:23:B6:2E and 00:07:62:0F:00:00 for stream: 1 using codec: SBC	Mar-31-2014 12:53:05.474864 PM
29	▲	0	🔗	N/A	13479	Packet retransmission for unknown CID.	Mar-31-2014 12:53:07.712976 PM
30	▲	1	🔗	N/A	14187	AVDTP packet loss detected based on missing packet sequence number.	Mar-31-2014 12:53:13.742943 PM
31	▲	1	🔗	N/A	14351	AVDTP packet loss detected based on missing packet sequence number.	Mar-31-2014 12:53:15.385434 PM

Figure 3.14 - Event Table

Several events can be selected by clicking and dragging over the events, or by holding down the Shift key and clicking on events. To select events that are not adjacent hold down the Ctrl key and click on the events.

When selecting multiple events, the Wave Panels will not scroll to the selected events.

The Event Table contains eight columns.

Table 3.6 - Event Table Columns

Name	Value	Description
Event	integer	System generated sequential numbering of events.
Severity		Information - provides information of interest but does not indicate a problem event.
		Warning - identifies a potential problem where further investigation may be appropriate
		Error - identifies a definite problem.
Stream Id	integer	A system generated ID that is assigned in the order that the audio streams are detected. The ID is not maintained between captures for the same device with the same audio. It identifies the Wave Panel where the event can be viewed. The ID appears in the Audio Stream Info of the Wave Panel.

Table 3.6 - Event Table Columns (continued)

Name	Value	Description
Event Type		Bluetooth -Events generated by analyzing Bluetooth protocol activities.
		Codec -Events generated from analyzing the audio coding/decoding activities.
		Audio -Events generated by analyzing the audio data.
Mode	N/A	Mode does not apply to this event.
	REF	Referenced Mode. Refer to 3.4 Referenced Mode Testing Processes on page 51 and 2.1.2 Referenced Mode Event Reporting on page 7 .
	UN-REF	Non-Referenced Mode. Refer to 1.1 Non-Referenced Mode on page 1 and 2.1.1 Non-Referenced Mode Event Reporting on page 3 .
Frame Number	integer	The system generated identification for a specific frame.
Description		Details and explanation about this event.
Timestamp	clock date and time	A system generated time stamp for each frame.

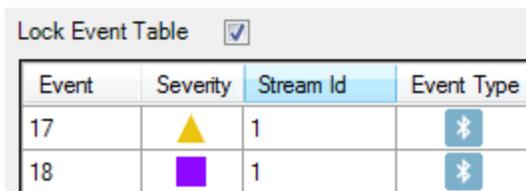
Sorting

Event table entries are sortable by column. Left-click on the column heading to sort.

Event Table Pop-Up Menu

Right-clicking with the cursor over the Event Table will open a menu of additional options. For more on this option see [Wave Panel & Event Table Pop-up Menu on page 46](#).

Lock Event Table

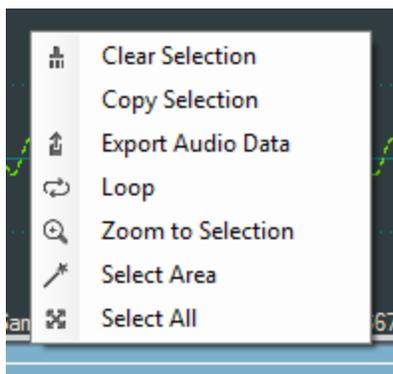


The **Lock Event Table** checkbox is available in live mode only. Clicking to check the box will prevent the Event Table from scrolling during live capture. Un-checking the box will resume scrolling of events as they are detected. When analyzing a capture file the checkbox has no effect.

3.2.4 Wave Panel & Event Table Pop-up Menu

Additional Wave Panel and Event Table options are available by right clicking the mouse with the cursor anywhere in the Wave Panel or in the Event Table.

Wave Panel Pop-up Menu Actions

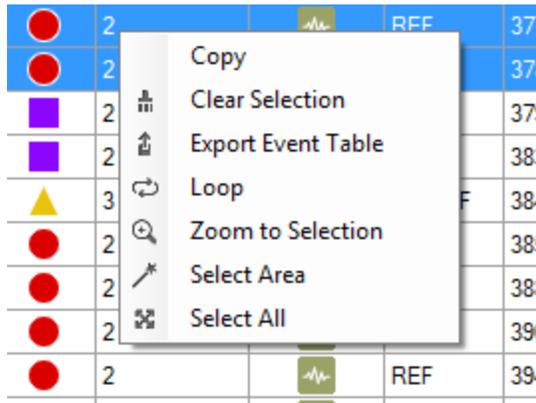


Right-clicking anywhere in the Wave Panel will provide you with a selection of the following actions.

Table 3.7 - Wave Panel Pop-up Menu Selections

Option	Description
Clear Selection	Clears the current selection in the viewer
Copy Selection	Saves a copy of the selection to the computer clipboard. The clipboard can be pasted into a Word document, an e-mail, or other Windows clipboard-compatible application.
Export Audio Data	Opens the Export pop-up menu with options to export the waveform as a .raw, .wav, or Event Data. For additional details on exporting refer to Waveform Display Export .
Loop	Loops through the audio selected on the Wave Panel.
Zoom to Selection	Expands or compresses the selection to fill the Wave Panel view.
Select Area	When the mouse cursor is positioned over data (not fill, pause, or gaps) in the Wave Panel and selecting this option will select all the data between and fills, pauses, or gaps.
Select All	Selects the entire waveform

Event Table Pop-up Menu Actions



Right-clicking in the Event Table will provide you with a selection of the following actions.

Table 3.8 - Event Table Pop-up Menu Selection

Options	Description
Copy	Copies the selected events to Windows clipboard as text.
Clear Selection	Clears the current event selection in the table
Export Event Table	Copies the current event selection and saves it as a .csv file. For additional details on exporting refer to Event Table Export .
Loop	Loops through the audio selected on the Wave Panel.
Zoom to Selection	Expands the Event Table selection to fill the Wave Panel view.
Select Area	Expands the selection.
Select All	Selects all events.

3.2.4.1 Export Audio Data

There are two ways to export audio data:

1. Clicking the Audio Expert System™ window **Global Toolbar** Export button .
2. Right-click in a Stream Panel Wave Panel and a pop-up menu will appear. Select **Export**.

Two windows will appear:

1. The standard Windows Save As.
2. The **Export Audio Data** dialog.

In the Windows Save As window enter a **File name** and directory location. Click on **Save**.

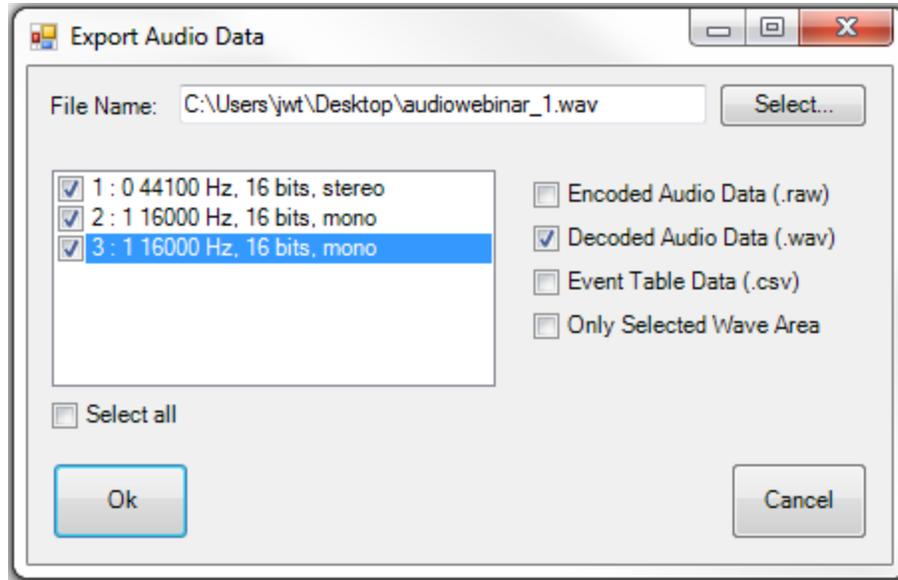


Figure 3.15 - Export Audio Data dialog

The Save As window will close, and the file name will appear in the **File Name** field in the **Export Audio Data** window. Should the file name need to be changed, click on the **Select** button and the Windows Save As dialog will open. By default the .wav file extension is used in the file name.

In the window below **File Name** will appear a list of **Stream IDs** with a description from the Audio Stream Info . If opening from the Audio Expert System™ **Global Toolbar** all **Stream IDs** are checked by default. If opening from a Wave Panel, the **Stream ID** where the export dialog was opened is automatically checked. You can check each stream that is to be exported. For convenience checking **Select all** below the stream list window will place checks in all streams.

Export Options

After selecting the streams to export, select the desired formats to export.

Table 3.9 - Export Audio Data Format Options

Option	Description
Encoded Audio Data	Exports the selected files as .raw format. The audio data is in an encrypted format and user will need a codec to decode it.
Decoded Audio Data	Exports the selected files as .wav format that can be played on a wide variety of media players.
Event Table Data	Exports a text .csv file of all the detected events
Only Selected Wave Area	Exports the Encoded, Decoded, or Event Data for the selected waveform. This option is only active if a selection has been made in one of the Wave Panels

	audiowebinar_1.csv	39 KB	Microsoft Excel C...
	audiowebinar_1.raw	5,439 KB	RAW File
	audiowebinar_1.wav	38,652 KB	Wave Sound
	audiowebinar_2.csv	39 KB	Microsoft Excel C...
	audiowebinar_2.raw	299 KB	RAW File
	audiowebinar_2.wav	7,227 KB	Wave Sound

Click on **OK** to save the waveform. The dialog will close and a series of progress bars will appear. Each progress bar is associated with a file for each export option. The exported files will have the following syntax: `<filename>_n.<filetype>`, where `<filename>` = the name entered into the File Name field, `n` = the stream id number (1, 2, 3, ...), and `<filetype>` = "raw", "wav", and "csv". The image shows an example where the user exported **Stream**

Id's 1 and 2 in Encoded Audio , Decoded Audio , and Event Table data to filename "audiowebinar".

Click on **Cancel** to close the window without exporting.

3.2.4.2 Export Event Table

Right-clicking in the Event table will open a pop-up menu with the option to **Export Event Table**. This option will export selected events in the in comma separated variable (.csv) format for used in Microsoft Excel or any other Windows .csv compatible application.

First select the events to export. Multiple events are selectable by selecting an event then holding the Shift key while clicking on another event. This will select all events between the two selections. If the selections are not adjacent you can hold the Ctrl (control) key while clicking events.

Next right-click anywhere in the Event Table to open the pop-up menu and click on the **Export Event Table** option. A Windows **Save As** dialog will open. Enter a file name and select a file location and click on **Save**. A confirmation dialog will open. Click **OK** to close the confirmation dialog.

If you have not selected an event in the table before exporting, a warning to "Please select an event row first." appears.

3.3 Frame, Packet, and Protocol Analysis Synchronization

The Audio Expert System™ module integrates seamlessly with Frontline software with common timestamping of *Bluetooth* protocol data, audio events, audio waveform display, and codec events. The audio expert data and results are synchronized and coordinated with the existing Frontline software data views, such as **Frame Display**, **Bluetooth Timeline**, etc. to expedite the root-cause analysis of *Bluetooth* protocol related audio issues. When a frame is selected in any Frontline software data views, the corresponding audio data associated with those frames is also selected in the Wave Panel, Event Timeline and Event Table and vice-verse.

Protocol analysis tools synchronized to the Audio Expert System™ include:

- **Frame Display**
- **Coexistence View**
- **Bluetooth Timeline**
- **Message Sequence Chart**
- **Packet Error Rate Statistics**

When a portion of the waveform is selected in the Wave Panel, all frames within the selection will be highlighted in the **Frame Display**, **Coexistence View**, and **Bluetooth Timeline**.

Note: If the **Frame Display** is filtered to show non-audio events then the frames associated with selected audio events may not show.

3.4 Referenced Mode Testing Processes

In the Referenced mode, the devices under test use a specific audio file (called reference file or test file) provided by Frontline whose contents are already known to the Frontline software. The software compares the parameters of the received audio data against its parameters and presents analysis for the user. Commonly, in Bluetooth technology the music sent via A2DP and speech sent via HFP. There are a few ways users can conduct referenced mode testing depending upon what profile they are using. The figure 17 shows the source of the audio and the medium through which it can be accessed by Source device to send to sink device via Bluetooth.

Table 3.10 - Referenced Mode Testing Process Between Two DUTs

Audio Source	Process to Send Using A2DP	Process to Send Using HFP
A file stored on the device's local memory	Play the locally stored file on the audio source device	Play using the third party App that transmits music data on HFP.
Streaming audio over a cellular network	Play the test in a browser on the audio source device https://youtu.be/mirDbikrtM	Make a call to 434-964-1407 or 434-964-1304 through a cellular network. The phone number receiving the call playbacks recorded test signal.
Streaming audio over a Wi-Fi network	Play the test in a browser on the audio source device https://youtu.be/mirDbikrtM	Make a call to 434-964-1407 or 434-964-1304 through a VoIP provider such as Skype. The phone number receiving the call playbacks recorded test signal. Potential problem: The VoIP provider might use custom codecs and cause undesirable behavior.

A2DP

Playing the test file locally

The simplest way to perform music data testing is to directly play the reference file from DUT1 to DUT2. To do that, save the reference file provided with the Frontline software on the Source device. Then connect the Bluetooth enabled devices and play the music file from one device to the other. The software will automatically detect the mode and present analysis for the user.

Playing the test file via Internet

If the user is testing a scenario where they need to analyze audio played through the internet (either using Wi-Fi or cellular data plan), they may access the reference file on YouTube provided by Frontline -

<https://youtu.be/rmirDbikrtM>. Note that the software is only analyzing the Bluetooth link between the two DUTs. Any abnormalities at the Wi-Fi and cellular network level will affect the audio quality that may not be Bluetooth protocol related and the software will not be able to detect that.

HFP

Playing the test file by calling a phone number

Frontline provides the following phone numbers - 434-964-1407 and 434-964-1304 that users can call, to conduct speech audio data analysis over Bluetooth. The calls can be made using the cellular network (most common method) or VoIP. Again, the VoIP provider might use custom codecs and cause undesirable behavior which cannot be detected by Audio Expert System™ software.

Playing the test file using Third party Apps

Bluetooth Audio Expert System™ Reference mode testing can be accomplished using third party apps on Android, iOS, and Windows phones. The following apps are available from their respective App stores:

- [BTmono, Android](#)
- [Blue2Car, IOS](#)
- [Windows Headset player lite](#)

Note: When selecting and using these apps, thoroughly review all the vendor documentation. While Teledyne LeCroy has conducted testing of these apps, Teledyne LeCroy has not completed full interoperability testing with our library of *Bluetooth* devices and does not warrant the use of these apps with every device when using the following procedures. Teledyne LeCroy does not provide support or maintenance for third party apps. Any issues or questions should be directed to the app developer.

1. In the following steps Device Under Test 1 (DUT1) is the device sending the reference test file to DUT2.
2. Download the third party app to DUT1 and follow the app vendor's instructions for installation and use.
3. Load the Audio Expert System reference test file

"Test_1.02_64.1kHz_16Bit.wav"

on DUT1. The test file is stored on the users computer In the directory "\\Frontline <version #>\Development Tools\Audio Expert Test Files\".

Note: Reference test files are periodically updated. Shown here is an example. Files delivered with your latest Frontline software version may have changed. Contact Teledyne LeCroy Technical Support for information on the latest reference file versions.

4. With the BPA 600 or Soderia connected to the computer, configure the datasource, and follow procedures to capture data.
5. Launch Audio Expert System by clicking on the **Control** window .
6. Turn on Bluetooth on your DUTs, DUT1 and DUT2. Turn on the third party *Bluetooth* app for routing the reference file over A2DP or HFP by following the vendor's directions.

7. Send the reference test file from DUT1 to DUT2 via the third party app.
8. Observe the events in the Audio Expert System™ **Events Table**. Look for an event **Description**:
 “TestIDFound : REF: Test ID 1.02, Channel Gain = -11.8 dB TermFreq=400.0”.

Note: This is an example. The display may vary with the reference file version.

The Frontline analyzer has successfully detected the reference test signal and the system is locked into reference mode.

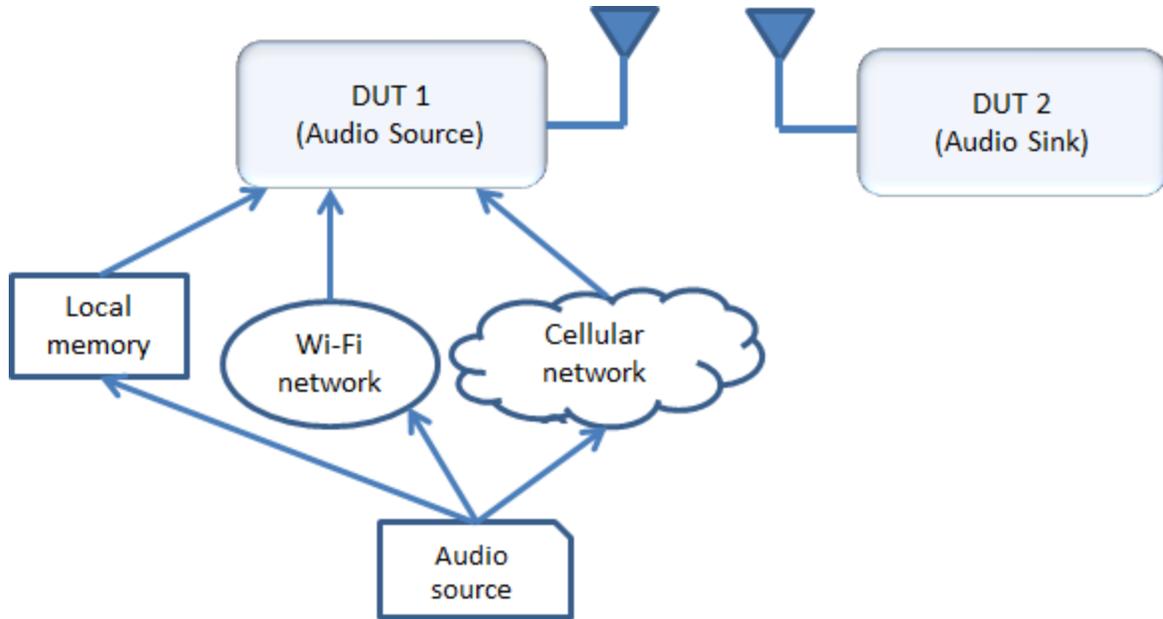


Figure 3.16 - Test Cases for Referenced Mode Testing

3.5 Air Sniffing: Positioning Devices

When capturing over the air packets, proper positioning of the Frontline hardware and the Devices Under Test (DUTs) will result in the best possible captures and will mitigate sources of path loss and interference. The following procedures will help optimize the capture process especially if you have problems obtaining reliable ...captures.

Problems with indoor radio propagation

Even in free space, it is well understood that radio frequencies attenuate over distance. The free-space rule-of-thumb dictates that radio energy decreases in strength by 20 dB by each 10-to-1 increase in range. In the real-world, the effects of objects in an outdoor environment cause reflection, diffraction, and scattering resulting in greater signal losses. Indoors the situation can be worse. Reflections occur from walls and other large flat surfaces. Diffraction occurs from objects with sharp edges. Scattering is produced from objects with rough surfaces and from small objects. Also any object directly in the path of the radiation can present a hard or soft partition depending on the partition's material properties. Path losses from partitions are difficult to estimate.

Estimating indoor propagation loss

One estimate of indoor path loss, based on path loss data from a typical building, provides a $\frac{1}{\text{range}^{3.5}}$ power rule. At 2.4 GHz, the following relationship provides an approximate estimate of indoor path loss:

$$\text{Indoor Path Loss (in dB)} = 40 + 35\text{Log}_{10}(\text{range, in meters})$$

This approximation is expected to have a variance of 13 dB.

Mitigating path loss and interference

Bluetooth device design contributes to mitigating environmental effects on propagation through spread spectrum radio design, for example. However, careful planning of the testing environment can also contribute to reliable data capture process.

The first step to ensuring reliable air-sniffing data capture is to understand the RF characteristics of the Devices Under Test (DUTs). The *Bluetooth* Class, antenna types, and radiation patterns are all important factors that can affect the placement of the DUTs and the Frontline hardware. Radiation patterns are rarely spherical, so understanding your device's radiation patterns can greatly enhance successful data capture. Position devices to avoid radiation attenuation by the surroundings.

This step is optional: Consider conductive testing to establish a baseline capture. Conductive testing isolates the DUTs and analyzer from environmental effects.

The next step is to ensure that the testing environment is as clutter-free as possible.

- Line-of-sight obstructions should be eliminated between the Frontline hardware and the DUTs because they cause a reduction in signal strength. Obstructions include, but are not limited to: water bottles, coffee cups, computers, computer screens, computer speakers, and books. A clear, unobstructed line-of-sight is preferred for DUT and Frontline hardware positioning.
- If using an analyzer connected to a computer, position the computer on an adjacent table or surface away from the analyzer and DUTs, taking advantage of the cables' length. If this is not possible, position the computer behind the analyzer as far away as possible. If using the Frontline FTS4BT, which is a dongle, either use an extension USB cable or position the computer such that the dongle is positioned towards the DUTs.
- The preferred placement is positioning the DUTs and the Frontline hardware at the points of an equilateral triangle in the same horizontal plane, i.e. placed on the same table or work surface. The sides of the triangle should be between 1 and 2 meters for *Bluetooth* transmitter classes 1 and 2. The distance for transmitter class 3 should be 1/2 meter.



Figure 3.17 - Devices Equally Spaced in the Same Horizontal Plane

Finally, eliminate other RF sources.

- Wi-Fi interference should be minimized or eliminated. *Bluetooth* shares the same 2.4 GHz frequency bands as Wi-Fi technology. Wi-Fi interference can cause loss of packets and poor captures. In a laboratory or testing environment do not place the DUTs and Frontline hardware in close proximity with Wi-Fi transmitting sources such as laptops or routers. Turning off Wi-Fi on the computer running the Frontline software is recommended.

Positioning for wideband capture

Frontline's Wideband Bluetooth Protocol Analyzer, Soderia, can capture from multiple devices, which requires a different approach to position the DUTs and the analyzer. When testing more than two devices arrange the DUTs on the perimeter of a circle 1-2 meters in diameter for Bluetooth transmitter Class 1 and 2 devices. For transmitter Class 3 DUTs, the circle should be 1/2 meter in diameter. Equally space the DUTs on the perimeter. Place the Soderia in the center of the circle. If not using the Soderia Excursion mode, connect the computer and place it outside the circle as far away from the DUTs as possible.

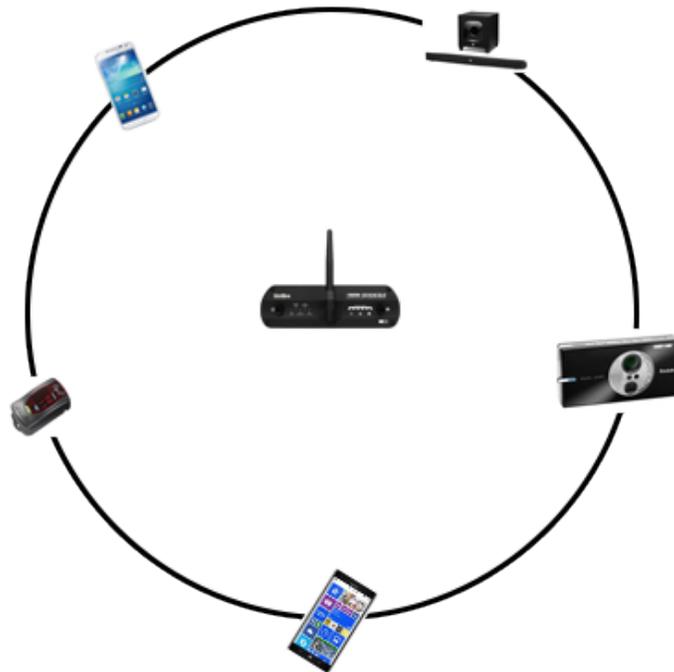


Figure 3.18 - Wideband Capture: Devices Equally Spaced in the Same Horizontal Plane

Positioning for audio capture

The Bluetooth Audio Expert System provides analysis of audio streams and can assist in identifying problems with capture methods including positioning and environment because it will point out missing frames. For hands-free profile data captures both DUTs send and receive data. Therefore, position the devices following the equilateral triangle arrangement as mentioned above.

However, in A2DP data capture scenario, the equilateral positioning of devices is not optimum because, normally, only one device is sending data to the other. It is recommended that the Frontline hardware be positioned closer to the device receiving data so that Frontline better mimics the receiving DUT. Position the DUTs 1 -2 meters apart for Class 1 and 2 transmitters, and 1/2 meter apart for Class 3 transmitters.



Figure 3.19 - For Audio A2DP, Position Closer to SINK DUT

Poor Placement

A poor test configuration for the analyzer is placing the DUTs very close to each other and the analyzer far away. The DUTs, being in close proximity to each other, reduce their transmission power and thus make it hard for the analyzer to hear the conversation. If the analyzer is far away from DUTs, there are chances that the analyzer may miss those frames, which could lead to failure in decryption of the data.

Obstacles in close proximity to or in between the analyzer and the DUTs can interfere and cause reduction in signal strength or interference. Even small objects can cause signal scattering.

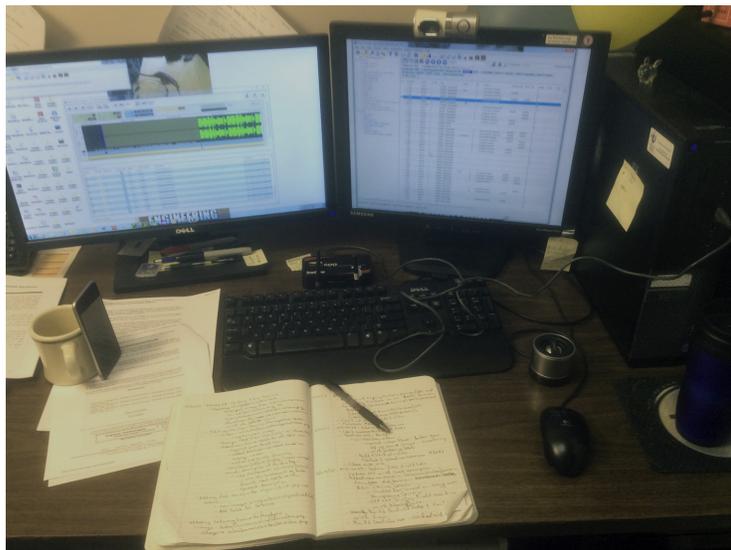


Figure 3.20 - Example: Poor Capture Environment

Contacting Technical Support

Technical support is available in several ways. The online help system provides answers to many user related questions. Frontline's website has documentation on common problems, as well as software upgrades and utilities to use with our products.

On the Web: <http://fte.com/support/supportrequest.aspx>

Email: tech_support@fte.com

If you need to talk to a technical support representative about your Audio Expert System product, support is available between 9 am and 5 pm, U.S. Eastern Time zone, and between 9 am and 5 pm, Pacific Time zone, on Monday through Friday. Technical support is not available on U.S. national holidays.

Phone: +1 (434) 984-4500

Fax: +1 (434) 984-4505

Instructional Videos

Teledyne LeCroy provides a series of videos to assist the user and may answer your questions. These videos can be accessed at fte.com/support/videos.aspx. On this web page use the **Video Filters** sidebar to select instructional videos for your product.