Extensible 3D (X3D)
Part 1: Architecture and base components

16 Sound component

16.1 Introduction

16.1.1 Name

The name of this component is "Sound". This name shall be used when referring to this component in the COMPONENT statement (see 7.2.5.4 Component statement).

16.1.2 Overview

This clause describes the Sound component of this part of ISO/IEC 19775. This includes how sound is delivered to an X3D world as well as how sounds are accessed. Table 16.1 provides links to the major topics in this clause.

Table 16.1 — Topics

16.2 Concepts

16.2.1 Audio and sound spatial architecture

The Sound component provides a rich set of spatialized audio capabilities in a comprehensive architecture suitable for 3D models and virtual environments.

a. *Signal sources for sound*. In addition to playing inputs from prerecorded sound files, capabilities are provided for computational audio generation, microphones, and virtual listening points.
b. *Virtual locations for sound generation*. The Sound and SpatialSound nodes define location, direction, and characteristics of expected sound production in virtual 3D space.
c. *Propagation*. Attenuation model may be further modified by AcousticProperties of reflection, refraction and absorption based on surrounding geometry.
d. *Reception points*. Avatar-centered listening points and recordable listening points, each with arbitrary location and direction, can receive acoustic results modified by corresponding with left-right pan and spatialization.

16.2.2 Sound priority

If the browser does not have the resources to play all of the currently active sounds, it is recommended that the browser sort the active sounds into an ordered list using the following sort keys in the order specified:

a. decreasing *priority;*
b. for sounds with *priority* > 0.5, increasing (now-*startTime*);
c. decreasing *intensity* at viewer location (*intensity* × "intensity attenuation");

where *priority* is the *priority* field of the Sound node, now represents the current time, *startTime* is the *startTime* field of the audio source node specified in the *source* field, and "intensity attenuation" refers to the intensity multiplier derived from the linear decibel attenuation ramp between inner and outer ellipsoids.

It is important that sort key 2 be used for the high priority (event and cue) sounds so that new cues are heard even when the browser is "full" of currently active high priority sounds. Sort key 2 should not be used for normal priority sounds, so selection among them is based on sort key 3 (intensity at the location of the viewer).

The browser shall play as many sounds from the beginning of this sorted list as it can given available resources and allowable latency between rendering. On most systems, the resources available for MIDI streams are different from those for playing sampled sounds, thus it may be beneficial to maintain a separate list to handle MIDI data.

16.2.3 Sound attenuation and spatialization

In order to create a linear decrease in loudness as the viewer moves from the inner to the outer ellipsoid of the sound, the attenuation must be based on a linear decibel ramp. To make the falloff consistent across browsers, the decibel ramp is to vary from 0 dB at the minimum ellipsoid to -20 dB at the outer ellipsoid. Sound nodes with an outer ellipsoid that is ten times larger than the minimum will display the inverse square intensity drop-off that approximates sound attenuation in an anechoic environment.

Browsers may support spatial localization of sounds whose *spatialize* field is TRUE as well as their underlying sound libraries will allow. Browsers shall at least support stereo panning of non-MIDI sounds based on the angle between the viewer and the source. This angle is obtained by projecting the Sound *location* (in global space) onto the XZ plane of the viewer. Determine the angle between the Z-axis and the vector from the viewer to the transformed *location*, and assign a pan value in the range [0.0, 1.0] as depicted in Figure 16.1. Given this pan value, left and right channel levels can be obtained using the following equations:

$$\text{leftPanFactor} = 1 - \text{pan}^2$$
$$\text{rightPanFactor} = 1 - (1 - \text{pan})^2$$

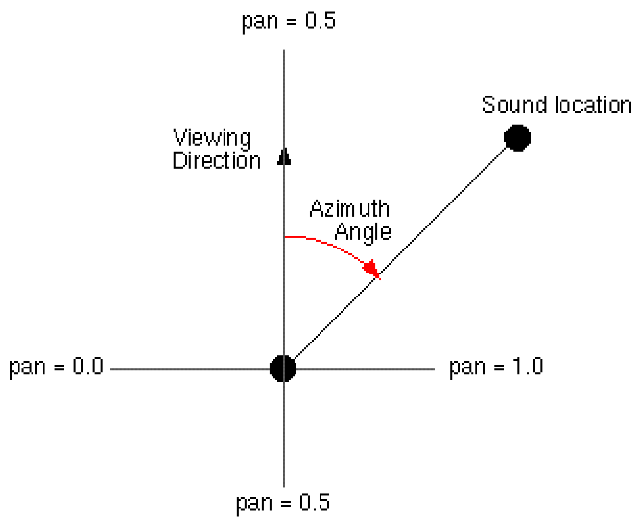


Figure 16.1 — Stereo panning

Using this technique, the loudness of the sound is modified by the *intensity* field value, then distance attenuation to obtain the unspatialized audio output. The values in the unspatialized audio output are then scaled by leftPanFactor and rightPanFactor to determine the final left and right output signals. The use of more sophisticated localization techniques is encouraged, but not required (see [SNDB]).

These planar gain-reduction relationships pertain to the location and relative direction of current avatar as well as any ListenerPoint nodes.

16.2.4 Sound propagation

Sound-propagation techniques can be used to simulate sound waves as they travel from each source to scene listening points by taking into account the expected interactions with various objects in the scene. In other words, spatial sound rendering includes the estimation of physical effects involved in sound propagation such as surface reflection (specular, diffusion) and wave phenomena (refraction, diffraction) within a 3D scene. Figure 16.2 provides an overview of the physical models of sound propagation that are considered.

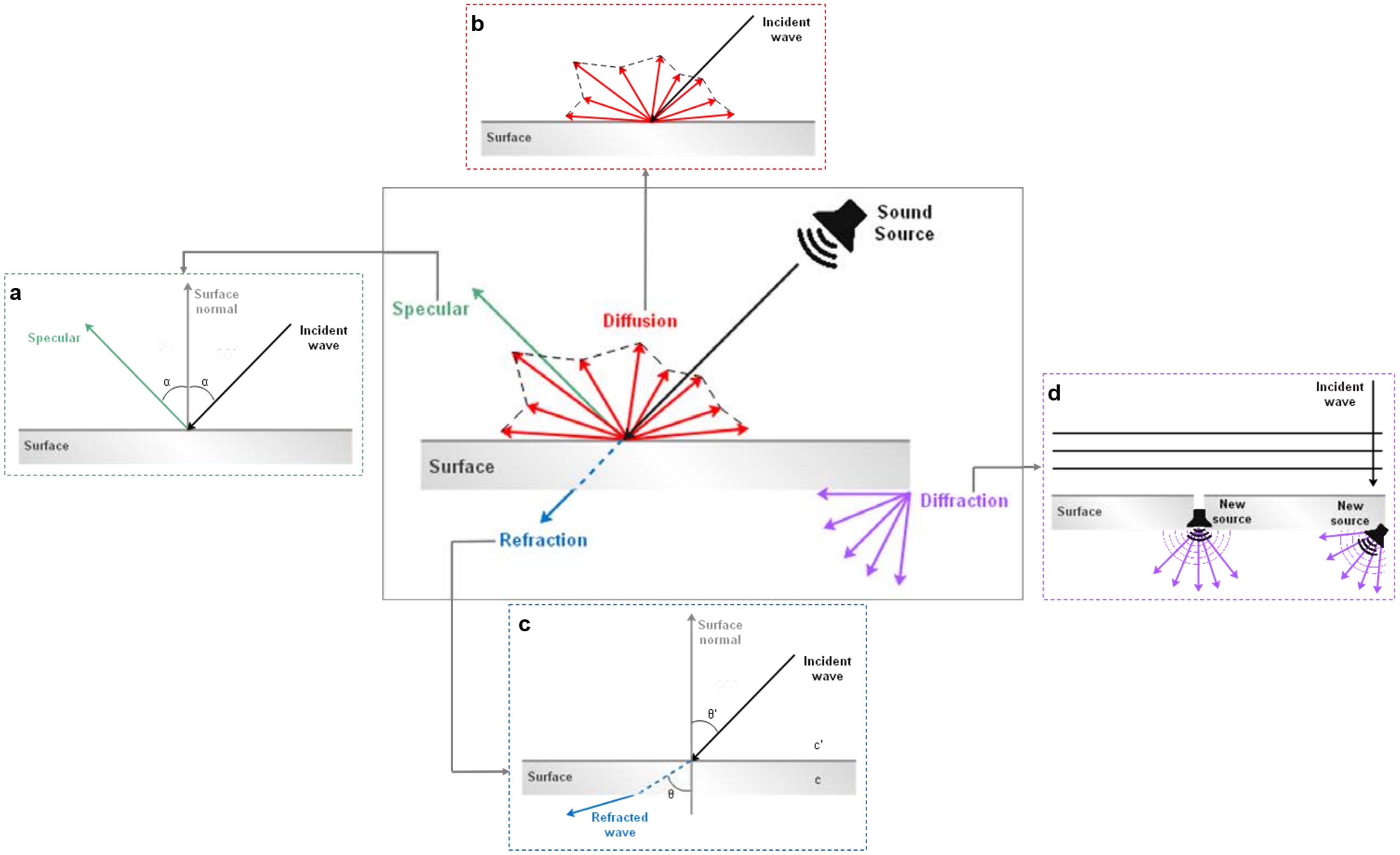


Figure 16.2 — Sound Propagation Phenomena

- *Specular and diffuse reflection:* during the propagation of a sound wave in an enclosed space, the wave hits objects or room boundaries and its free propagation is disturbed. Moreover, during this process, at least a portion of the incident wave is thrown back, a phenomenon known as reflection. If the wavelength of the sound wave is small enough with respect to the dimensions of the reflecting object and large compared with possible irregularities of the reflecting surface, a specular reflection occurs. This phenomenon is illustrated in Figure 16.2 (inset a), in which the angle of reflection is equal to the angle of incidence. In contrast, if the sound wavelength is comparable to the corrugation dimensions of an irregular reflection surface, the incident sound wave is scattered in many directions. In this case, the phenomenon is called diffuse reflection and is illustrated in Figure 16.2 (inset b).
- *Refraction:* it is the change in the propagation direction of waves when they obliquely cross the boundary between two mediums where their speed changes, as shown in Figure 16.2 (inset c). For transmission of a plane sound wave from air into another medium, the refraction index in following equation (Snell's Law) is used, for calculating the geometric conditions.
  `n = c'/c = sinθ'/sinθ` where c′ and c the sound speed in the two media, θ the angle of incidence and θ′ the angle of refraction.
- *Diffraction:* the fact that a listener can hear sounds around corners and around barriers involves a diffraction model of sound. It is the spread of waves around corners, behind obstacles or around the edges of an opening as illustrated in Figure 16.2 (inset d). The amount of diffraction increases with wavelength, meaning that sound waves with lower frequencies, and thus with greater wavelengths than obstacles or openings dimensions, is spread over larger regions behind the openings or around the obstacles.

(TODO: consider improvement or removal.) Diffraction sources are not explicitly represented in this component, and often can be handled by computational engines. Complex geometric openings may also be modeled by an audio chain including ListenerPoint and SpatialSound to emulate sophisticated diffraction propagation paths.

If a simplified geometry alternative from Collision *proxy* field is available, it is used preferentially by collision-detection algorithms for sound propagation, rather than descendant children of the Collision node. Such geometric simplifications can often reduce computational costs significantly without reduction in perceived audio fidelity of 3D scene acoustics. (TODO: consider need for *acousticProxy* field, or if Shape/Appearance/AcousticProperties is sufficient.)

16.2.5 Sound effects processing

Sound streams can be manipulated by a variety of sound effects. Audio graphs are a powerful mechanism for modeling the diversity of real-world and electronic modifications to sound that can occur. Close integration of sound rendering and effects with 3D models and aggregate scenes provides powerful capabilities for increased realism.

Historically a wide variety of computational libraries for sound generation and propagation have been available, often with significant differences and limitations. Sound propagation and effects processing in this component are based on design patterns found in W3C Web Audio API [W3C-WebAudio]. Design goals of that specification include supporting "the capabilities found in modern game audio engines as well as some of the mixing, processing, and filtering tasks that are found in modern desktop audio production applications." These capabilities are broad, implemented in a variety of libraries, and deployed in multiple Web browsers. The primary interfaces of W3C Web Audio API [W3C-WebAudio] necessary for creating audio graphs have corresponding X3D node support in this component.

TODO continued design, implementation and evaluation work for this component is needed to ensure that full coverage of W3C Audio API capabilities is achieved.

Descriptions follow for a number of fields that are common to multiple nodes related to sound processing.

The *channelCount* field is the number of channels used when up-mixing and down-mixing connections to any inputs of a node. The default value is typically 2 except for specific nodes where its value is specially determined This attribute has no effect for nodes with no inputs.

The *channelCountMode* field is used to determine the *computedNumberOfChannels* that controls how inputs to a node are to be mixed.

- `"max"`: use *computedNumberOfChannels* (value for *channelCount* is ignored)
- `"clamped-max"`: use *computedNumberOfChannels* clamped to maximum value given by *channelCount*
- `"explicit"`: Up-mix by filling channels until they run out then zero out remaining channels. Down-mix by filling as many channels as possible, then dropping remaining channels. .

The *channelInterpretation* field determines how individual channels are treated when up-mixing and down-mixing connections to any inputs to the node. The default value is "speakers". This attribute has no effect for nodes with no inputs. Allowed values include the following:

- `"speakers"`: use up-mix equations or down-mix equations. In cases where the number of channels do not match any of these basic speaker layouts, revert to `"discrete"`.
- `"discrete"`: *computedNumberOfChannels* is the exact value as specified by the *channelCount*.

The *gain* field is a factor that represents the amount of linear amplification to apply to the output of the node. Decibel values shall not be used. Negative gain factors negate the input signal.

The *numberOfInputs* field is the number of inputs feeding into a node.

The *numberOfOutputs* field is the number of outputs coming out of a node.

16.3 Abstract types

16.3.1 *X3DSoundAnalysisNode*

```
X3DSoundAnalysisNode : X3DNode {
  SFString [in,out] description  ""
  SFBool   [in,out] enabled      TRUE
  SFFloat  [in,out] gain         1 [-∞,∞)

  SFInt32  [~~in,~~out] channelCount          0         [0,∞)
  SFString [in,out] channelCountMode      "max"     ["max", "clamped-max", "explicit"]
  SFString [in,out] channelInterpretation "speakers" ["speakers", "discrete"]
  ~~SFInt32  [in,out] numberOfInputs        0         [0,∞)~~
  ~~SFInt32  [in,out] numberOfOutputs       0         [0,∞)~~
  # Mechanisms for parent-child input-output graph design remain under review
  SFNode   [in out] inputs                NULL      [X3DSoundAnalysisNode,X3DSoundChannelNode,X3DSoundProcessingNode,X3DSoundSourceNode]
}
```

This is the base node type for nodes which receive real-time generated data, without any change from the input to output sound information.

TODO: if enabled FALSE, does signal pass through unmodified or is it blocked? Perhaps an additional boolean is needed for pass-through state? Modeling the 'connect' attribute and defining defaults is necessary for each case.

16.3.2 *X3DSoundChannelNode*

```
X3DSoundChannelNode : X3DTimeDependentNode {
  SFString [in,out] description  ""
  SFBool   [in,out] enabled      TRUE
  SFFloat  [in,out] gain         1     [-∞,∞)
  SFBool   [in,out] loop         FALSE
  SFNode   [in,out] metadata     NULL  [X3DMetadataObject]
  SFTime   [in,out] pauseTime    0     (-∞,∞)
  SFTime   [in,out] resumeTime   0     (-∞,∞)
  SFTime   [in,out] startTime    0     (-∞,∞)
  SFTime   [in,out] stopTime     0     (-∞,∞)
  SFTime   [out]    elapsedTime
  SFBool   [out]    isActive
  SFBool   [out]    isPaused

  SFInt32  [~~in,~~out] channelCount          0         [0,∞)
  SFString [in,out] channelCountMode      "max"     ["max", "clamped-max", "explicit"]
  SFString [in,out] channelInterpretation "speakers" ["speakers", "discrete"]
  ~~SFInt32  [in,out] numberOfInputs        0         [0,∞)~~
  ~~SFInt32  [in,out] numberOfOutputs       0         [0,∞)~~
  # Mechanisms for parent-child input-output graph design remain under review
  MFNode   [in out] inputs                NULL      [X3DSoundAnalysisNode,X3DSoundChannelNode,X3DSoundProcessingNode,X3DSoundSourceNode]
}
```

This is the base node type for nodes that handle of channels in an audio stream, allowing them to be split or merged.

(Section moved here and adapted from AudioClip.)
The *description, enabled, loop, pauseTime, resumeTime, startTime,* and *stopTime* inputOutput fields and the *elapsedTime, isActive,* and *isPaused* outputOnly fields, and their effects on nodes implementing this abstract node type, are discussed in detail in X3DTimeDependentNode and 8.2.4 Time-dependent nodes.

TODO: if enabled FALSE, does signal pass through unmodified or is it blocked? Perhaps an additional boolean is needed for pass-through state? Modeling the 'connect' attribute and defining defaults is necessary for each case.

16.3.3 *X3DSoundDestinationNode*

```
X3DSoundDestinationNode : X3DTimeDependentNode {
  SFString [in,out] description  ""
  SFFloat  [in,out] gain         1     [-∞,∞)
  SFBool   [in,out] enabled      TRUE
  SFBool   [in,out] loop         FALSE
  SFNode   [in,out] metadata     NULL  [X3DMetadataObject]
  SFTime   [in,out] pauseTime    0     (-∞,∞)
  SFTime   [in,out] resumeTime   0     (-∞,∞)
  SFTime   [in,out] startTime    0     (-∞,∞)
  SFTime   [in,out] stopTime     0     (-∞,∞)
  SFTime   [out]    elapsedTime
  SFBool   [out]    isActive
  SFBool   [out]    isPaused

  SFInt32  [~~in,~~out] channelCount          0          [0,∞)
  SFString [in,out] channelCountMode      "max"      ["max", "clamped-max", "explicit"]
  SFString [in,out] channelInterpretation "speakers" ["speakers", "discrete"]
  ~~SFInt32  [in,out] numberOfInputs        0          [0,∞)~~
  ~~SFInt32  [in,out] numberOfOutputs       0          [0,∞)~~
  # Mechanisms for parent-child input-output graph design remain under review
  MFNode   [in out] inputs                NULL       [X3DSoundAnalysisNode,X3DSoundChannelNode,X3DSoundProcessingNode,X3DSoundSourceNode]
}
```

This is the base node type for all sound destination nodes, which represent the final destination of an audio signal and are what the user can ultimately hear. Such nodes are often considered as audio output devices which are connected to speakers. All rendered audio that is intended to be heard gets routed to these terminal nodes.

(Section moved here and adapted from AudioClip.)
The *description, enabled, loop, pauseTime, resumeTime, startTime,* and *stopTime* inputOutput fields and the *elapsedTime, isActive,* and *isPaused* outputOnly fields, and their effects on nodes implementing this abstract node type, are discussed in detail in X3DTimeDependentNode and 8.2.4 Time-dependent nodes.

TODO: if enabled FALSE, does signal pass through unmodified or is it blocked? Perhaps an additional boolean is needed for pass-through state? Modeling the 'connect' attribute and defining defaults is necessary for each case.

16.3.4 *X3DSoundProcessingNode*

```
X3DSoundProcessingNode : X3DTimeDependentNode {
  SFString [in,out] description  ""
  SFBool   [in,out] enabled      TRUE
  SFFloat  [in,out] gain         1     [-∞,∞)
  SFBool   [in,out] loop         FALSE
  SFNode   [in,out] metadata     NULL  [X3DMetadataObject]
  SFTime   [in,out] pauseTime    0     (-∞,∞)
  SFTime   [in,out] resumeTime   0     (-∞,∞)
  SFTime   [in,out] startTime    0     (-∞,∞)
  SFTime   [in,out] stopTime     0     (-∞,∞)
  SFTime   [out]    elapsedTime
  SFBool   [out]    isActive
  SFBool   [out]    isPaused
  # Mechanisms for parent-child input-output graph design remain under review
  MFNode   [in out] inputs               NULL       [X3DSoundAnalysisNode,X3DSoundChannelNode,X3DSoundProcessingNode,X3DSoundSourceNode]
}
```

This is the base node type for all sound processing nodes, which are used to enhance audio with filtering, delaying, changing gain, etc.

(Section moved here and adapted from AudioClip.)
The *description, enabled, loop, pauseTime, resumeTime, startTime,* and *stopTime* inputOutput fields and the *elapsedTime, isActive,* and *isPaused* outputOnly fields, and their effects on nodes implementing this abstract node type, are discussed in detail in X3DTimeDependentNode and 8.2.4 Time-dependent nodes.

TODO: if enabled FALSE, does signal pass through unmodified or is it blocked? Perhaps an additional boolean is needed for pass-through state? Modeling the 'connect' attribute and defining defaults is necessary for each case.

16.3.5 *X3DSoundNode*

```
X3DSoundNode : X3DChildNode {
  SFString [in,out] description   ""
  SFFloat  [in,out] gain          1    [-∞,∞)
  SFNode   [in,out] metadata      NULL [X3DMetadataObject]
}
```

This abstract node type is the base for all sound nodes.

16.3.6 *X3DSoundSourceNode*

```
X3DSoundSourceNode : X3DTimeDependentNode {
  SFString [in,out] description      ""
  SFFloat  [in,out] gain             1    [-∞,∞)
  SFBool   [in,out] loop             FALSE
  SFNode   [in,out] metadata         NULL [X3DMetadataObject]
  SFTime   [in,out] pauseTime        0    (-∞,∞)
  SFFloat  [in,out] pitch            1.0  (0,∞)
  SFTime   [in,out] resumeTime       0    (-∞,∞)
  SFTime   [in,out] startTime        0    (-∞,∞)
  SFTime   [in,out] stopTime         0    (-∞,∞)
  SFTime   [out]    duration_changed
  SFTime   [out]    elapsedTime
  SFBool   [out]    isActive
  SFBool   [out]    isPaused
  # Mechanisms for parent-child input-output graph design remain under review
  MFNode   [in out] inputs            NULL      [X3DSoundAnalysisNode,X3DSoundChannelNode,X3DSoundProcessingNode,X3DSoundSourceNode]
}
```

This abstract node type is used to derive node types that can emit audio data.

(Section moved here and adapted from AudioClip.)
The *description, loop, pauseTime, resumeTime, startTime,* and *stopTime* inputOutput fields and the *elapsedTime, isActive,* and *isPaused* outputOnly fields, and their effects on nodes implementing this abstract node type, are discussed in detail in X3DTimeDependentNode and 8.2.4 Time-dependent nodes.

The *pitch* field specifies a multiplier for the rate at which sampled sound is played. Values for the *pitch* field shall be greater than zero. Changing the *pitch* field affects both the pitch and playback speed of a sound. A *set_pitch* event to an active ~~AudioClip~~ node is ignored and no *pitch_changed* field is generated. If *pitch* is set to 2.0, the sound shall be played one octave higher than normal and played twice as fast. For a sampled sound, the *pitch* field alters the sampling rate at which the sound is played. The proper implementation of pitch control for MIDI (or other note sequence sound clips) is to multiply the tempo of the playback by the *pitch* value and adjust the MIDI Coarse Tune and Fine Tune controls to achieve the proper pitch change.

A *duration_changed* event is sent whenever there is a new value for the "normal" duration of the clip. Typically, this will only occur when the current *url* in use changes and the sound data has been loaded, indicating that the clip is playing a different sound source. The duration is the length of time in seconds for one cycle of the audio for a *pitch* set to 1.0. Changing the *pitch* field will not trigger a *duration_changed* event. A duration value of "−1" implies that the sound data has not yet loaded or the value is unavailable for some reason. A *duration_changed* event shall be generated if the ~~AudioClip~~ node is loaded when the X3D file is read or the ~~AudioClip~~ node is added to the scene graph.

The *isActive* field may be used by other nodes to determine if the ~~clip~~ node is currently active.

16.4 Node reference

16.4.1 Analyser

```
Analyser : X3DSoundAnalysisNode {
  SFString [in,out] description          ""
  SFBool   [in,out] enabled              TRUE
  SFInt32  [in,out] fftSize              2048 [0,∞)
  SFInt32  [in,out] frequencyBinCount    1024 [0,∞)
  SFFloat  [in,out] gain                 1    [-∞,∞)
  SFFloat  [in,out] minDecibels          -100 (-∞,∞)
  SFFloat  [in,out] maxDecibels          -30  (-∞,∞)
```

```
    SFFloat  [in,out] smoothingTimeConstant 0.8  [0,∞)

    SFInt32  [~~in,~~out] channelCount          0          [0,∞)
    SFString [in,out] channelCountMode      "max"      ["max", "clamped-max", "explicit"]
    SFString [in,out] channelInterpretation "speakers" ["speakers", "discrete"]
    ~~SFInt32  [in,out] numberOfInputs        0          [0,∞)~~
    ~~SFInt32  [in,out] numberOfOutputs       0          [0,∞)~~
    # Mechanisms for parent-child input-output graph design remain under review
    MFNode   [in out] inputs                NULL       [X3DSoundAnalysisNode,X3DSoundChannelNode,X3DSoundProcessingNode,X3DSoundSourceNode]
  }
```

The Analyser node provides real-time frequency and time-domain analysis information, without any change to the input.

The *fftSize* field is an unsigned long value representing the size of the FFT (Fast Fourier Transform) to be used to determine the frequency domain.

The *frequencyBinCount* field is an unsigned long value half that of the FFT size. This generally equates to the number of data values you will have to play with for the visualization.

The *minDecibels* field is a value representing the minimum power value in the scaling range for the FFT analysis data, for conversion to unsigned byte values.

The *maxDecibels* field is a value representing the maximum power value in the scaling range for the FFT analysis data, for conversion to unsigned byte values.

The *smoothingTimeConstant* field is a value representing the averaging constant with the last analysis frame.

TODO determine if accessType is outputOnly for derived information

16.4.2 AudioBufferSource

```
  AudioBufferSource : X3DSoundSourceNode {
    MFFloat  [in,out] buffer                NULL [-1,1]
    SFString [in,out] description           ""
    SFFloat  [in,out] detune                0    [0,∞)
    SFFloat  [in,out] duration              0    [0,∞)
    SFFloat  [in,out] gain                  1    [-∞,∞)
    SFBool   [in,out] loop                  FALSE
    SFFloat  [in,out] loopStart             0    [0,∞)
    SFFloat  [in,out] loopEnd               0    [0,∞)
    SFInt32  [in,out] numberOfChannels      0    [0,∞)
    SFFloat  [in,out] playbackRate          0    [-∞,∞)
    SFFloat  [in,out] sampleRate            0    [0,∞)
    SFInt32  [out]    length                0    [0,∞)

    SFInt32  [~~in,~~out] channelCount          0          [0,∞)
    SFString [in,out] channelCountMode      "max"      ["max", "clamped-max", "explicit"]
    SFString [in,out] channelInterpretation "speakers" ["speakers", "discrete"]
    ~~SFInt32  [in,out] numberOfInputs        0          [0,∞)~~
    ~~SFInt32  [in,out] numberOfOutputs        0           [0,∞)~~
    # Mechanisms for parent-child input-output graph design remain under review
  }
```

The AudioBufferSource node represents a memory-resident audio asset. Its format is non-interleaved 32-bit floating-point linear PCM values with a normal range of [−1,1], but values are not limited to this range. It can contain one or more channels. Typically, it would be expected that the length of the PCM data would be fairly short (usually somewhat less than a minute). For longer sounds, such as music soundtracks, streaming such as StreamAudioSource should be used.

The *buffer* field is a data block holding the audio sample data.

The *detune* field

The *duration* field indicates the duration of the PCM audio data in seconds, computed from the *length* field divided by *sampleRate* field.

The *length* field is the length of the PCM audio data in sample-frames.

The *numberOfChannels* field is the discrete number of audio channels for this buffer.

The *playbackRate* field is the speed at which to render the audio stream.

The *sampleRate* field is the sample-rate used for the PCM audio data in samples per second.

16.4.3 AudioClip

```
AudioClip : X3DSoundSourceNode, X3DUrlObject {
  SFString [in,out] description           ""
  SFFloat  [in,out] gain                  1    [-∞,∞)
  SFBool   [in,out] load             TRUE
  SFBool   [in,out] loop             FALSE
  SFNode   [in,out] metadata         NULL  [X3DMetadataObject]
  SFTime   [in,out] pauseTime        0     (-∞,∞)
  SFFloat  [in,out] pitch            1.0   (0,∞)
  SFTime   [in,out] refresh          0.0   [0,∞)
  SFTime   [in,out] resumeTime       0     (-∞,∞)
  SFTime   [in,out] startTime        0     (-∞,∞)
  SFTime   [in,out] stopTime         0     (-∞,∞)
  MFString [in,out] url              []    [URI]
  SFTime   [out]    duration_changed
  SFTime   [out]    elapsedTime
  SFBool   [out]    isActive
  SFBool   [out]    isPaused
}
```

An AudioClip node specifies audio data that can be referenced by Sound nodes.

The *description* field specifies a textual description of the audio source. A browser is not required to display the *description* field but may choose to do so in addition to playing the sound.

The *url* field specifies the URL from which the sound is loaded. Browsers shall support at least the *wavefile* format in uncompressed PCM format (see [WAV]). It is recommended that browsers also support the MIDI file type 1 sound format (see 2.[MIDI]) and the MP3 compressed format (see 2.[I11172-1]). MIDI files are presumed to use the General MIDI patch set. 9.2.1 URLs contains details on the *url* field.

(Sections moved to parent interface X3DSoundSourceNode and related interfaces in this component.)
~~The *loop, pauseTime, resumeTime, startTime,* and *stopTime* inputOutput fields and the *elapsedTime, isActive,* and *isPaused* outputOnly fields, and their effects on the AudioClip node, are discussed in detail in 8 Time component.~~

~~The *pitch* field specifies a multiplier for the rate at which sampled sound is played. Values for the *pitch* field shall be greater than zero. Changing the *pitch* field affects both the pitch and playback speed of a sound. A *set_pitch* event to an active AudioClip is ignored and no *pitch_changed* field is generated. If *pitch* is set to 2.0, the sound shall be played one octave higher than normal and played twice as fast. For a sampled sound, the *pitch* field alters the sampling rate at which the sound is played. The proper implementation of pitch control for MIDI (or other note sequence sound clips) is to multiply the tempo of the playback by the *pitch* value and adjust the MIDI Coarse Tune and Fine Tune controls to achieve the proper pitch change.~~

~~A *duration_changed* event is sent whenever there is a new value for the "normal" duration of the clip. Typically, this will only occur when the current *url* in use changes and the sound data has been loaded, indicating that the clip is playing a different sound source. The duration is the length of time in seconds for one cycle of the audio for a *pitch* set to 1.0. Changing the *pitch* field will not trigger a *duration_changed* event. A duration value of "−1" implies that the sound data has not yet loaded or the value is unavailable for some reason. A *duration_changed* event shall be generated if the AudioClip node is loaded when the X3D file is read or the AudioClip node is added to the scene graph.~~

The "*cycle*" of an AudioClip is the length of time in seconds for one playing of the audio at the specified *pitch*.

The *isActive* field may be used by other nodes to determine if the clip is currently active. If an AudioClip is active, it shall be playing the sound corresponding to the sound time (*i.e.*, in the sound's local time system with sample 0 at time 0):

```
t = (now - startTime) modulo (duration / pitch)
```

16.4.4 AudioDestination

```
AudioDestination : X3DSoundDestinationNode {
  SFString [in,out] description          ""
  SFFloat  [in,out] gain                 1          [-∞,∞)
  SFInt32  [in,out] maxChannelCount      2          [0,∞)
```

```
    SFInt32  [~~in,~~out] channelCount          0          [0,∞)
    SFString [in,out] channelCountMode      "max"      ["max", "clamped-max", "explicit"]
    SFString [in,out] channelInterpretation "speakers" ["speakers", "discrete"]
    ~~SFInt32  [in,out] numberOfInputs        0          [0,∞)~~
    ~~SFInt32  [in,out] numberOfOutputs       0          [0,∞)~~
    # Mechanisms for parent-child input-output graph design remain under review
    MFNode   [in out] inputs                NULL       [X3DSoundAnalysisNode,X3DSoundChannelNode,X3DSoundProcessingNode,X3DSoundSourceNode]
  }
```

AudioDestination represents the final audio destination and is what user ultimately hears, typically from the speakers of user device. An AudioDestinationNode representing the audio hardware end-point (the normal case) can potentially output more than 2 channels of audio if the audio hardware is multi-channel.

The *maxChannelCount* field is the maximum number of channels that the destination is capable of supporting.

16.4.5 BiquadFilter

```
  BiquadFilter : X3DSoundProcessingNode {
    SFString [in,out] description           ""
    SFFloat  [in,out] detune                0          [0,∞)
    SFInt32  [in,out] frequency             350        [0,∞)
    SFFloat  [in,out] Q                     1          [0,∞)
    SFFloat  [in,out] gain                  0          [0,∞)
    SFString [in,out] type                  "lowpass"  ["lowpass",   "highpass", "bandpass", "lowshelf",
                                                        "highshelf", "peaking",  "notch",    "allpass"]

    SFInt32  [~~in,~~out] channelCount          0          [0,∞)
    SFString [in,out] channelCountMode      "max"      ["max", "clamped-max", "explicit"]
    SFString [in,out] channelInterpretation "speakers" ["speakers", "discrete"]
    ~~SFInt32  [in,out] numberOfInputs        0          [0,∞)~~
    ~~SFInt32  [in,out] numberOfOutputs       0          [0,∞)~~
    # Mechanisms for parent-child input-output graph design remain under review
    MFNode   [in out] inputs                NULL       [X3DSoundAnalysisNode,X3DSoundChannelNode,X3DSoundProcessingNode,X3DSoundSourceNode]
  }
```

BiquadFilter represents different kinds of filters, tone control devices, and graphic equalizers. Low-order filters are the building blocks of basic tone controls (bass, mid, treble), graphic equalizers, and more advanced filters. Multiple BiquadFilterNode filters can be combined to form more complex filters. The filter parameters such as frequency can be changed over time for filter sweeps, etc.

The *detune* field is a detune value, in cents, for the frequency..

The *frequency* field is the frequency at which the BiquadFilterNode will operate, in Hz.

The *gain* field is the amplitude gain of the filter. Its value is in dB units. The *gain* value is only used for lowshelf, highshelf, and peaking filters.

The *Q* field is Quality Factor (Q) of the filter.

The *type* field is the type of this BiquadFilterNode. Note that the meaning of the different properties (*frequency*, detune and *Q*) differs depending on the type of the filter used.

Enumeration	Description
`"lowpass"`	A lowpass filter allows frequencies below the cutoff frequency to pass through and attenuates frequencies above the cutoff. It implements a standard second-order resonant lowpass filter with 12dB/octave rolloff. frequency The cutoff frequency Q Controls how peaked the response will be at the cutoff frequency. A large value makes the response more peaked. gain Not used in this filter type
`"highpass"`	A highpass filter is the opposite of a lowpass filter. Frequencies above the cutoff frequency are passed through, but frequencies below the cutoff are attenuated. It implements a standard second-order resonant highpass filter with 12dB/octave rolloff. frequency The cutoff frequency below which the frequencies are attenuated Q Controls how peaked the response will be at the cutoff frequency. A large value makes the response more peaked. gain Not used in this filter type
`"bandpass"`	A bandpass filter allows a range of frequencies to pass through and attenuates the frequencies below and above this frequency range. It implements a second-order bandpass filter. frequency

	The center of the frequency band Q Controls the width of the band. The width becomes narrower as the Q value increases. gain Not used in this filter type
"lowshelf"	The lowshelf filter allows all frequencies through, but adds a boost (or attenuation) to the lower frequencies. It implements a second-order lowshelf filter. frequency The upper limit of the frequences where the boost (or attenuation) is applied. Q Not used in this filter type. gain The boost, in dB, to be applied. If the value is negative, the frequencies are attenuated.
"highshelf"	The highshelf filter is the opposite of the lowshelf filter and allows all frequencies through, but adds a boost to the higher frequencies. It implements a second-order highshelf filter frequency The lower limit of the frequences where the boost (or attenuation) is applied. Q Not used in this filter type. gain The boost, in dB, to be applied. If the value is negative, the frequencies are attenuated.
"peaking"	The peaking filter allows all frequencies through, but adds a boost (or attenuation) to a range of frequencies. frequency The center frequency of where the boost is applied.

	Q Controls the width of the band of frequencies that are boosted. A large value implies a narrow width. gain The boost, in dB, to be applied. If the value is negative, the frequencies are attenuated.
`"notch"`	The notch filter (also known as a band-stop or band-rejection filter) is the opposite of a bandpass filter. It allows all frequencies through, except for a set of frequencies. frequency The center frequency of where the notch is applied. Q Controls the width of the band of frequencies that are attenuated. A large value implies a narrow width. gain Not used in this filter type.
`"allpass"`	An allpass filter allows all frequencies through, but changes the phase relationship between the various frequencies. It implements a second-order allpass filter frequency The frequency where the center of the phase transition occurs. Viewed another way, this is the frequency with maximal group delay. Q Controls how sharp the phase transition is at the center frequency. A larger value implies a sharper transition and a larger group delay. gain Not used in this filter type.

16.4.6 ChannelMerger

```
ChannelMerger : X3DSoundChannelNode {
  SFString [in,out] description  ""
  SFBool   [in,out] enabled      TRUE
  SFFloat  [in,out] gain         1     [0,∞)
  SFBool   [in,out] loop         FALSE
  SFNode   [in,out] metadata     NULL  [X3DMetadataObject]
  SFTime   [in,out] pauseTime    0     (-∞,∞)
  SFTime   [in,out] resumeTime   0     (-∞,∞)
  SFTime   [in,out] startTime    0     (-∞,∞)
  SFTime   [in,out] stopTime     0     (-∞,∞)
  SFTime   [out]    elapsedTime
  SFBool   [out]    isActive
  SFBool   [out]    isPaused

  SFInt32  [~~in,~~out] channelCount          0          [0,∞)
  SFString [in,out] channelCountMode      "max"      ["max", "clamped-max", "explicit"]
  SFString [in,out] channelInterpretation "speakers" ["speakers", "discrete"]
  ~~SFInt32  [in,out] numberOfInputs        0          [0,∞)~~
  ~~SFInt32  [in,out] numberOfOutputs       0          [0,∞)~~
  # Mechanisms for parent-child input-output graph design remain under review
  MFNode   [in out] inputs                NULL       [X3DSoundAnalysisNode,X3DSoundChannelNode,X3DSoundProcessingNode,X3DSoundSourceNode]
}
```

ChannelMerger unites different input channels into a single output channel.

16.4.7 ChannelSelector

```
ChannelSelector : X3DSoundChannelNode {
  SFString [in,out] description  ""
  SFBool   [in,out] enabled      TRUE
  SFFloat  [in,out] gain         1     [0,∞)
  SFBool   [in,out] loop         FALSE
  SFNode   [in,out] metadata     NULL  [X3DMetadataObject]
  SFTime   [in,out] pauseTime    0     (-∞,∞)
  SFTime   [in,out] resumeTime   0     (-∞,∞)
  SFTime   [in,out] startTime    0     (-∞,∞)
  SFTime   [in,out] stopTime     0     (-∞,∞)
  SFTime   [out]    elapsedTime
  SFBool   [out]    isActive
  SFBool   [out]    isPaused

  SFInt32  [in,out] channelNumber         0          [0,∞)

  SFInt32  [~~in,~~out] channelCount          0          [0,∞)
  SFString [in,out] channelCountMode      "max"      ["max", "clamped-max", "explicit"]
  SFString [in,out] channelInterpretation "speakers" ["speakers", "discrete"]
  ~~SFInt32  [in,out] numberOfInputs        0          [0,∞)~~
  ~~SFInt32  [in,out] numberOfOutputs       0          [0,∞)~~
  # Mechanisms for parent-child input-output graph design remain under review
  MFNode   [in out] inputs                NULL       [X3DSoundAnalysisNode,X3DSoundChannelNode,X3DSoundProcessingNode,X3DSoundSourceNode]
}
```

ChannelSelector selects a single channel output from all input channels. If the selected channel is not monophonic, further splitting remains feasible.

The *channelNumber* field indicates which channel to select, with index values beginning at 0.

16.4.8 ChannelSplitter

```
ChannelSplitter : X3DSoundChannelNode {
  SFString [in,out] description  ""
  SFBool   [in,out] enabled      TRUE
  SFFloat  [in,out] gain         1     [0,∞)
  SFBool   [in,out] loop         FALSE
  SFNode   [in,out] metadata     NULL  [X3DMetadataObject]
  SFTime   [in,out] pauseTime    0     (-∞,∞)
  SFTime   [in,out] resumeTime   0     (-∞,∞)
  SFTime   [in,out] startTime    0     (-∞,∞)
  SFTime   [in,out] stopTime     0     (-∞,∞)
  SFTime   [out]    elapsedTime
  SFBool   [out]    isActive
  SFBool   [out]    isPaused

  SFInt32  [~~in,~~out] channelCount          0          [0,∞)
  SFString [in,out] channelCountMode      "max"      ["max", "clamped-max", "explicit"]
  SFString [in,out] channelInterpretation "speakers" ["speakers", "discrete"]
  ~~SFInt32  [in,out] numberOfInputs        0          [0,∞)~~
  ~~SFInt32  [in,out] numberOfOutputs       0          [0,∞)~~
  # Mechanisms for parent-child input-output graph design remain under review
  SFNode   [in out] inputs                NULL       [X3DSoundAnalysisNode,X3DSoundChannelNode,X3DSoundProcessingNode,X3DSoundSourceNode]
  MFNode   [in out] outputs               NULL       [X3DSoundAnalysisNode,X3DSoundChannelNode,X3DSoundProcessingNode,X3DSoundSourceNode]
}
```

ChannelSplitter separates the different channels of an audio source into a set of monophonic output channels.

16.4.9 Convolver

```
Convolver : X3DSoundProcessingNode {
  MFFloat  [in,out] buffer                NULL [-1,1]
  SFString [in,out] description           ""
  SFFloat  [in,out] gain                  1          [0,∞)
  SFBool   [in,out] normalize             FALSE

  SFInt32  [~~in,~~out] channelCount          0          [0,∞)
  SFString [in,out] channelCountMode      "max"      ["max", "clamped-max", "explicit"]
  SFString [in,out] channelInterpretation "speakers" ["speakers", "discrete"]
  ~~SFInt32  [in,out] numberOfInputs        0          [0,∞)~~
  ~~SFInt32  [in,out] numberOfOutputs       0          [0,∞)~~
  # Mechanisms for parent-child input-output graph design remain under review
  MFNode   [in out] inputs                NULL       [X3DSoundAnalysisNode,X3DSoundChannelNode,X3DSoundProcessingNode,X3DSoundSourceNode]
}
```

Convolver performs a linear convolution on a given AudioBuffer, often used to achieve a reverberation effect. Potential modifications include chorus effects, reverberation, and telephone-like speech.

The idea for producing room effects is to play back a reference sound in a room, record it, and then (metaphorically) take the difference between the original sound and the recorded one. The result of this is an impulse response that captures the effect that the room has on a sound. These impulse responses are painstakingly recorded in very specific studio settings, and doing this on your own requires serious dedication. There are sites that host many of these pre-recorded impulse response files (stored as audio files). The Web Audio API provides an easy way to apply these impulse responses to your sounds using the ConvolverNode.

The *buffer* field represents a memory-resident audio asset (for one-shot sounds and other short audio clips). Its format is non-interleaved 32-bit linear floating-point PCM values with a normal range of [−1,1], but values are not limited to this range. It can contain one or more channels. Typically, it would be expected that the length of the PCM data would be fairly short (usually somewhat less than a minute). For longer sounds, such as music soundtracks, streaming should be used with the `<audio>` HTML element and AudioClip.

The *normalize* field is a boolean that controls whether the impulse response from the buffer is scaled by an equal-power normalization when the buffer attribute is set, or not.

16.4.10 Delay

```
Delay : X3DSoundProcessingNode {
  SFInt32  [in,out] delayTime             0          [0,∞)
  SFString [in,out] description           ""
  SFFloat  [in,out] gain                  1          [0,∞)

  SFInt32  [~~in,~~out] channelCount          0          [0,∞)
  SFString [in,out] channelCountMode      "max"      ["max", "clamped-max", "explicit"]
  SFString [in,out] channelInterpretation "speakers" ["speakers", "discrete"]
  ~~SFInt32  [in,out] numberOfInputs        0          [0,∞)~~
  ~~SFInt32  [in,out] numberOfOutputs       0          [0,∞)~~
  # Mechanisms for parent-child input-output graph design remain under review
  MFNode   [in out] inputs                NULL       [X3DSoundAnalysisNode,X3DSoundChannelNode,X3DSoundProcessingNode,X3DSoundSourceNode]
}
```

Delay causes a time delay between the arrival of input data and subsequent propagation to the output.

The *delayTime* field represents the amount of delay (in seconds) to apply.

16.4.11 DynamicsCompressor

```
DynamicsCompressor : X3DSoundProcessingNode {
  SFFloat  [in,out] attack                0.003      [0,∞)
  SFString [in,out] description           ""
  SFFloat  [in,out] gain                  1          [0,∞)
  SFInt32  [in,out] knee                  30         [0,∞)
  SFInt32  [in,out] ratio                 12         [0,∞)
  SFFloat  [in,out] reduction             0          [0,∞)
  SFInt32  [in,out] release               0.25       (-∞,∞)
  SFFloat  [in,out] threshold             -24        [0,∞)

  SFInt32  [~~in,~~out] channelCount          0          [0,∞)
  SFString [in,out] channelCountMode      "max"      ["max", "clamped-max", "explicit"]
  SFString [in,out] channelInterpretation "speakers" ["speakers", "discrete"]
  ~~SFInt32  [in,out] numberOfInputs         0          [0,∞)~~
  ~~SFInt32  [in,out] numberOfOutputs        0          [0,∞)~~
  # Mechanisms for parent-child input-output graph design remain under review
  MFNode   [in out] inputs                 NULL       [X3DSoundAnalysisNode,X3DSoundChannelNode,X3DSoundProcessingNode,X3DSoundSourceNode]
}
```

DynamicsCompressor implements a dynamics compression effect, lowering the volume of the loudest parts of the signal and raises the volume of the softest parts.

The *attack* field is the amount of time (in seconds) to reduce the gain by 10dB.

The *knee* field contains a decibel value representing the range above the threshold where the curve smoothly transitions to the compressed portion.

The *ratio* field represents the amount of change, in dB, needed in the input for a 1 dB change in the output.

The *reduction* field represents the amount of gain reduction in dB currently applied by the compressor to the signal. If fed no signal, then the value will be 0 (no gain reduction).

The *release* field represents the amount of time (in seconds) to increase the gain by 10dB.

The *threshold* field represents the decibel value above which the compression will start taking effect.

16.4.12 Gain

```
Gain : X3DSoundProcessingNode {
  SFString [in,out] description            ""
  SFFloat  [in,out] gain                   1          [-∞,∞)

  SFInt32  [~~in,~~out] channelCount           0          [0,∞)
  SFString [in,out] channelCountMode       "max"      ["max", "clamped-max", "explicit"]
  SFString [in,out] channelInterpretation "speakers" ["speakers", "discrete"]
  ~~SFInt32  [in,out] numberOfInputs         0          [0,∞)~~
  ~~SFInt32  [in,out] numberOfOutputs        0          [0,∞)~~
  # Mechanisms for parent-child input-output graph design remain under review
  MFNode   [in out] inputs                 NULL       [X3DSoundAnalysisNode,X3DSoundChannelNode,X3DSoundProcessingNode,X3DSoundSourceNode]
}
```

The Gain node amplifies or deamplifies the input signal.

The *gain* field is a factor that represents the amount of linear amplification to apply. Decibel values shall not be used. Negative gain factors negate the input signal.

16.4.13 ListenerPoint

```
ListenerPoint : X3DAudioListenerNode {
  SFBool     [in]     set_bind
```

```
    SFString   [in,out] description        ""
    SFBool     [in,out] enabled            TRUE
    SFInt32    [in,out] gain               1       [0,∞)
    SFFloat    [in out] interauralDistance 0       [0, infinity)
    SFNode     [in,out] metadata           NULL    [X3DMetadataObject]
    SFRotation [in,out] orientation        0 0 1 0 [-1,1],(-∞,∞)
    SFVec3f    [in,out] position           0 0 10  (-∞,∞)
    SFBool     [in,out] trackCurrentView   FALSE
    SFTime     [out]    bindTime
    SFBool     [out]    isBound
  # Mechanisms for parent-child input-output graph design remain under review
  }
```

ListenerPoint represents the position and orientation of the person listening to the audio scene. It provides single or multiple sound channels as output. Multiple ListenerPoint nodes can be active for sound processing, but only one can be bound as the active listening point for the user.

The *interauralDistance* field is used for binaural recording.

If TRUE the *trackCurrentView* field matches *position* and *orientation* to the user's current view.

16.4.14 MicrophoneSource

```
  MicrophoneSource : X3DSoundSourceNode {
    SFString [in,out] description    ""
    SFBool   [in,out] enabled        TRUE
    SFFloat  [in,out] gain           1     [0,∞)
    SFBool   [in,out] isActive       FALSE
    SFString [in,out] mediaDeviceID  ""
  # Mechanisms for parent-child input-output graph design remain under review
  }
```

MicrophoneSource captures input from a physical microphone.

The *mediaDeviceID* field is a unique identifier for the represented device.

TODO: reconcile whether all the many fields of X3DSoundSourceNode are appropriate.

16.4.15 OscillatorSource

```
  Oscillator : X3DSoundSourceNode {
    SFString [in,out] description      ""
    SFFloat  [in,out] gain             1      [0,∞)
    SFBool   [in,out] loop             FALSE
    SFNode   [in,out] metadata         NULL   [X3DMetadataObject]
    SFTime   [in,out] pauseTime        0      (-∞,∞)
    SFFloat  [in,out] pitch            1.0    (0,∞)
    SFTime   [in,out] resumeTime       0      (-∞,∞)
    SFTime   [in,out] startTime        0      (-∞,∞)
    SFTime   [in,out] stopTime         0      (-∞,∞)
    SFTime   [out]    duration_changed
    SFTime   [out]    elapsedTime
    SFBool   [out]    isActive
    SFBool   [out]    isPaused

    SFFloat  [in,out] detune           0      [0,∞)
    SFInt32  [in,out] frequency        0      [0,∞)
    SFNode   [in,out] periodicWave     NULL   [PeriodicWave]
    SFString [in,out] type             "sine" ["sine", "square", "sawtooth", "triangle", "custom"]
    # Mechanisms for parent-child input-output graph design remain under review
  }
```

The Oscillator node represents an audio source generating a periodic waveform, providing a constant tone.

The *detune* field is an a-rate AudioParam representing detuning of oscillation in cents (though the AudioParam returned is read-only, the value it represents is not).

The *frequency* field is an a-rate AudioParam representing the frequency of oscillation in hertz (though the AudioParam returned is read-only, the value it represents is not). The default value is 440 Hz (a standard middle-A note).

The *periodicWave* field is an PeriodicWave used when type="custom" is indicated.

The *type* field is a string which specifies the shape of waveform to play; this can be one of a number of standard values, or custom to use a PeriodicWave to describe a custom waveform. Different types of waves produce different sounds. Standard values are "sine", "square", "sawtooth", "triangle" and "custom". Allowed values are

- `"sine"`: a sine wave
- `"square"`: a square wave of duty period 0.5
- `"sawtooth"`: a sawtooth wave
- `"triangle"`: a triangle wave
- `"custom"`: a custom periodic wave

16.4.16 PeriodicWave

```
  PeriodicWave : X3DSoundProcessingNode {
    SFString  [in,out] description ""
    SFFloat   [in,out] gain        1        [0,∞)
    SFInt32   [in,out] frequency   0        [0,∞)
    SFString  [in,out] type        "square"
    SFFloat   [in,out] detune      0        [0,∞)
  }
```

PeriodicWave defines a periodic waveform that can be used to shape the output of an Oscillator.

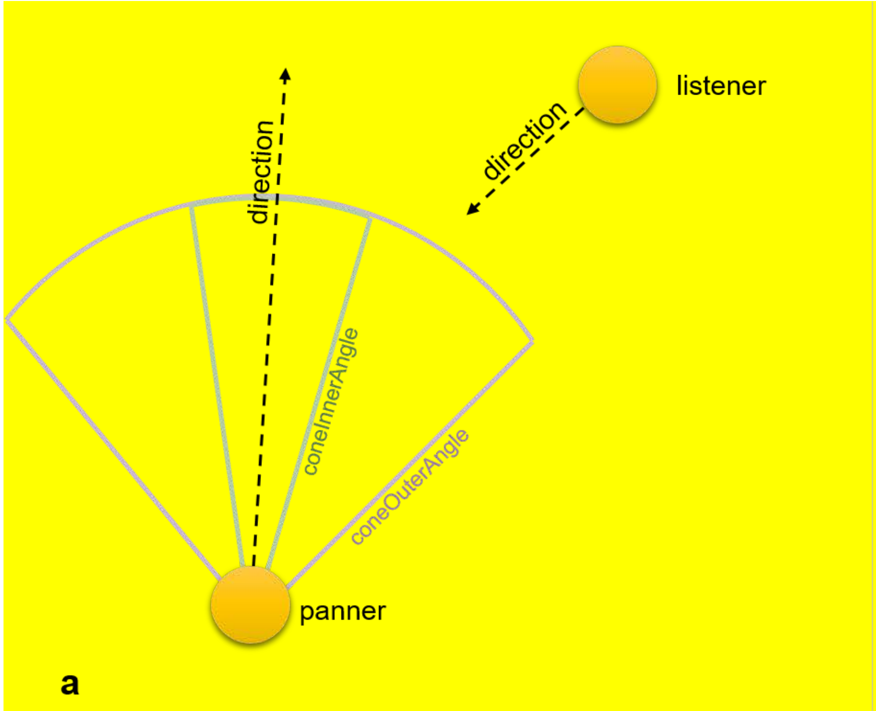
TODO confirm and describe attributes

16.4.17 *SpatialSound*

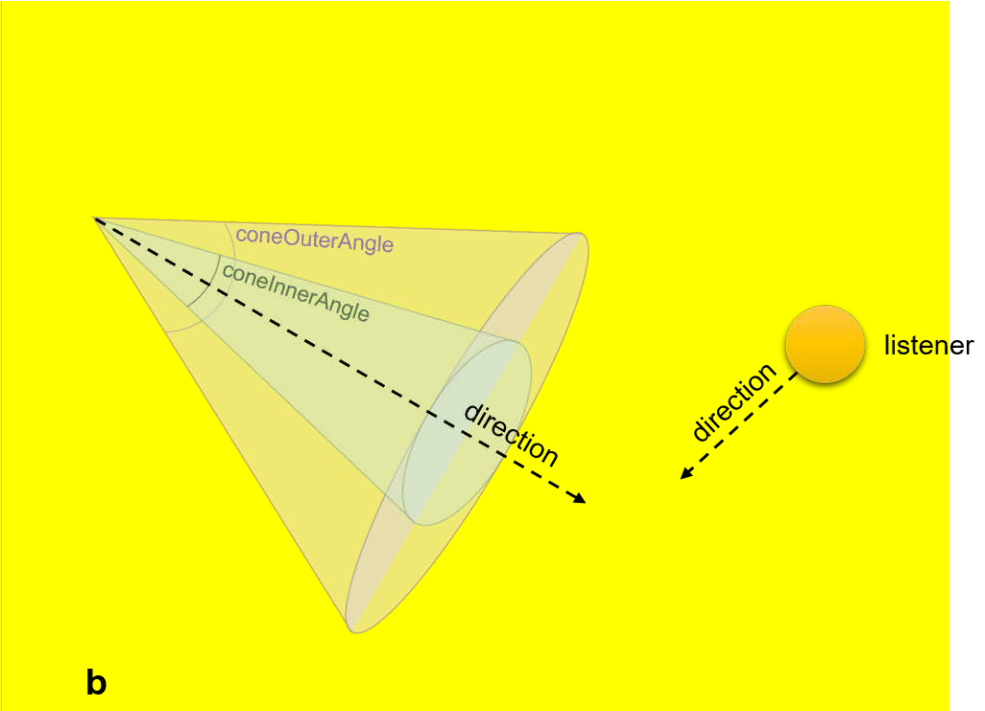
```
SpatialSound : X3DSoundNode  {
  SFFloat  [in,out] coneInnerAngle    6.2832    [0,2π]
  SFFloat  [in,out] coneOuterAngle    6.2832    [0,2π]
  SFFloat  [in,out] coneOuterGain     0         (-∞,∞)
  SFString [in,out] description       ""
  SFVec3f  [in,out] direction         0 0 1     (-∞,∞)
  SFString [in,out] distanceModel     "INVERSE" ["LINEAR" "INVERSE" "EXPONENTIAL"]
  SFFloat  [in,out] gain              1         [0,∞)
  SFFloat  [in,out] intensity         1         [0,1]
  SFVec3f  [in,out] location          0 0 0     (-∞,∞)
  SFFloat  [in,out] maxDistance       10000     [0,∞)
  SFNode   [in,out] metadata          NULL      [X3DMetadataObject]
  SFBool   [in,out] enableHRTF        FALSE
  SFFloat  [in,out] referenceDistance 1         [0,∞)
  SFFloat  [in,out] rolloffFactor     1         [0,∞)
  SFFloat  [in,out] priority          0         [0,1]
  SFNode   [in,out] source            NULL      [X3DSoundSourceNode] # and other types
  SFBool   []       spatialize        TRUE
  # Mechanisms for parent-child input-output graph design remain under review
  MFNode   [in out] inputs            NULL      [X3DSoundAnalysisNode,X3DSoundChannelNode,X3DSoundProcessingNode,X3DSoundSourceNode]
}
```

SpatialSound represents a processing node which positions, emits and spatializes an audio stream in three-dimensional space.

The *coneInnerAngle* is centered along direction and defines the inner conical volume, inside of which no source gain reduction occurs. The *coneOuterAngle* is centered along direction and defines an outer conical volume, within which the sound gain decreases linearly from full gain to *coneOuterGain*. Outside of *coneOuterAngle*, gain equals *coneOuterGain*. The value of *coneOuterAngle* is greater than or equal to *coneInnerAngle*. Corresponding gain reductions for 2D and 3D spatial panning between this source and a viewer (or ListenerPoint) are shown in Figure 16.3.



A diagram of panner and the listener in 2D space



A diagram of panner and the listener in 3D space

Figure 16.3 — SpatialSound Panning Gain Relationships for viewer (or ListenerPoint)

The *direction*. *intensity*, *location*, *priority*, *source* and *spatialize* fields match field definitions for Sound node.

The *referenceDistance* field is reference distance for reducing volume as source moves further from the listener.

The *rolloffFactor* field indicates how quickly volume is reduced as source moves further from listener.

The *distanceModel* field specifies which algorithm to use for sound attenuation, corresponding to distance between an audio source and a listener. as it moves away from the listener.

a. LINEAR gain model determined by
   `1 - rolloffFactor * (distance - referenceDistance) / (maxDistance - referenceDistance)`
b. INVERSE gain model determined by
   `refDistance / (referenceDistance + rolloffFactor * (Math.max(distance, referenceDistance) - referenceDistance))`
c. EXPONENTIAL gain model determined by
   `pow((Math.max(distance, referenceDistance) / referenceDistance, -rolloffFactor)`

The *enableHRTF* field specifies whether to enable Head Related Transfer Function (HRTF) auralization, if available.

The *maxDistance* field is the maximum distance where sound is renderable between source and listener, after which no reduction in sound volume occurs.

Spatial sound has a conceptual role in the Web3D environments, due to highly realism scenes that can provide. Since Web Audio API is the most popular sound engine, we propose to get the necessary steps required to make X3D fully compatible with this library. In fact, we propose the enrichment of X3D with spatial sound features, using the structure and the functionality of Web Audio API.

Particularly, the Web Audio API involves handling audio operations inside an audio context and has been designed to allow modular routing. Also, the approach of Web Audio API is based on the concept of audio context, which represents the direction of audio stream flows between sound nodes.

16.4.18 Sound

```
Sound : X3DSoundNode {
  SFVec3f [in,out] direction  0 0 1 (-∞,∞)
  SFString [in,out] description       ""
  SFFloat [in,out] gain       1     [0,∞)
  SFFloat [in,out] intensity  1     [0,1]
  SFVec3f [in,out] location   0 0 0 (-∞,∞)
  SFFloat [in,out] maxBack    10    [0,∞)
  SFFloat [in,out] maxFront   10    [0,∞)
  SFNode  [in,out] metadata   NULL  [X3DMetadataObject]
  SFFloat [in,out] minBack    1     [0,∞)
  SFFloat [in,out] minFront   1     [0,∞)
  SFFloat [in,out] priority   0     [0,1]
  SFNode  [in,out] source     NULL  [X3DSoundSourceNode]
  SFBool  []       spatialize TRUE
  # Mechanisms for parent-child input-output graph design remain under review
  MFNode   [in out] inputs            NULL      [X3DSoundAnalysisNode,X3DSoundChannelNode,X3DSoundProcessingNode,X3DSoundSourceNode]
}
```

The Sound node specifies the spatial presentation of a sound in a X3D scene. The sound is located at a point in the local coordinate system and emits sound in an elliptical pattern (defined by two ellipsoids). The ellipsoids are oriented in a direction specified by the *direction* field. The shape of the ellipsoids may be modified to provide more or less directional focus from the location of the sound.

The *source* field specifies the sound source for the Sound node. If the *source* field is not specified, the Sound node will not emit audio. The *source* field shall specify either an AudioClip node or a MovieTexture node. If a MovieTexture node is specified as the sound source, the MovieTexture shall refer to a movie format that supports sound (EXAMPLE MPEG-1Systems, see ISO/IEC 11172-1).

The *intensity* field adjusts the loudness (decibels) of the sound emitted by the Sound node. The *intensity* field has a value that ranges from 0.0 to 1.0 and specifies a factor which shall be used to scale the normalized sample data of the sound source during playback. A Sound node with an intensity of 1.0 shall emit audio at its maximum loudness (before attenuation), and a Sound node with an intensity of 0.0 shall emit no audio. Between these values, the loudness should increase linearly from a -20 dB change approaching an *intensity* of 0.0 to a 0 dB change at an *intensity* of 1.0.

NOTE This is different from the traditional definition of intensity with respect to sound; see [SNDA].

The *priority* field provides a hint for the browser to choose which sounds to play when there are more active Sound nodes than can be played at once due to either limited system resources or system load. 16.2 Concepts describes a recommended algorithm for determining which sounds to play under such circumstances. The *priority* field ranges from 0.0 to 1.0, with 1.0 being the highest priority and 0.0 the lowest priority.

The *location* field determines the location of the sound emitter in the local coordinate system. A Sound node's output is audible only if it is part of the traversed scene. Sound nodes that are descended from LOD, Switch, or any grouping or prototype node that disables traversal (*i.e.*, drawing) of its children are not audible unless they are traversed. If a Sound node is disabled by a Switch or LOD node, and later it becomes part of the traversal again, the sound shall resume where it would have been had it been playing continuously.

The Sound node has an inner ellipsoid that defines a volume of space in which the maximum level of the sound is audible. Within this ellipsoid, the normalized sample data is scaled by the *intensity* field and there is no attenuation. The inner ellipsoid is defined by extending the *direction* vector through the *location*. The *minBack* and *minFront* fields specify distances behind and in front of the *location* along the *direction* vector respectively. The inner ellipsoid has one of its foci at *location* (the second focus is implicit) and intersects the *direction* vector at *minBack* and *minFront*.

The Sound node has an outer ellipsoid that defines a volume of space that bounds the audibility of the sound. No sound can be heard outside of this outer ellipsoid. The outer ellipsoid is defined by extending the *direction* vector through the *location*. The *maxBack* and *maxFront* fields specify distances behind and in front of the *location* along the *direction* vector respectively. The outer ellipsoid has one of its foci at *location* (the second focus is implicit) and intersects the *direction* vector at *maxBack* and *maxFront*.

The *minFront*, *maxFront*, *minBack*, and *maxBack* fields are defined in local coordinates, and shall be greater than or equal to zero. The *minBack* field shall be less than or equal to *maxBack*, and *minFront* shall be less than or equal to *maxFront.* The ellipsoid parameters are specified in the local coordinate system but the ellipsoids' geometry is affected by ancestors' transformations.

Between the two ellipsoids, there shall be a linear attenuation ramp in loudness, from 0 dB at the minimum ellipsoid to -20 dB at the maximum ellipsoid:

```
attenuation = -20 × (d' / d")
```

where d' is the distance along the location-to-viewer vector, measured from the transformed minimum ellipsoid boundary to the viewer, and d" is the distance along the location-to-viewer vector from the transformed minimum ellipsoid boundary to the transformed maximum ellipsoid boundary (see Figure 16.2).

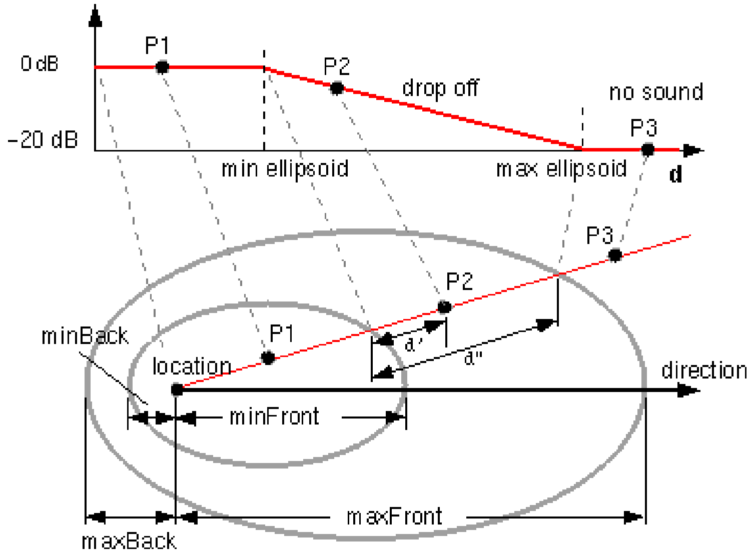


Figure 16.2 — Sound Node Geometry

The *spatialize* field specifies if the sound is perceived as being directionally located relative to the viewer. If the *spatialize* field is TRUE and the viewer is located between the transformed inner and outer ellipsoids, the viewer's direction and the relative location of the Sound node should be taken into account during playback. Details outlining the minimum required spatialization functionality can be found in 16.2.3 Sound attenuation and spatialization. If the *spatialize* field is FALSE, directional effects are ignored, but the ellipsoid dimensions and *intensity* will still affect the loudness of the sound. If the sound source is multi-channel (EXAMPLE stereo), the source shall retain its channel separation during playback.

16.4.19 StreamAudioDestination

```
StreamAudioDestination : X3DSoundDestinationNode {
  SFString [in,out] description           ""
  SFFloat  [in,out] gain                  1          [0,∞)
  MFFloat  [in,out] stream                NULL [-1,1]

  SFInt32  [~~in,~~out] channelCount          0          [0,∞)
  SFString [in,out] channelCountMode      "max"      ["max", "clamped-max", "explicit"]
  SFString [in,out] channelInterpretation "speakers" ["speakers", "discrete"]
  ~~SFInt32  [in,out] numberOfInputs        0          [0,∞)~~
  ~~SFInt32  [in,out] numberOfOutputs       0          [0,∞)~~
  # Mechanisms for parent-child input-output graph design remain under review
  MFNode   [in out] inputs                NULL       [X3DSoundAnalysisNode,X3DSoundChannelNode,X3DSoundProcessingNode,X3DSoundSourceNode]
}
```

StreamAudioDestination is an audio destination representing a MediaStream with a single MediaStreamTrack whose kind is "audio".

TODO confirm and describe attributes

16.4.20 StreamAudioSource

```
StreamAudioSource : X3DSoundSourceNode {
  SFString [in,out] description           ""
  SFFloat  [in,out] gain                  1          [0,∞)
  MFFloat  [in,out] mediaStream           []         [-1,1]

  SFInt32  [~~in,~~out] channelCount          0          [0,∞)
  SFString [in,out] channelCountMode      "max"      ["max", "clamped-max", "explicit"]
  SFString [in,out] channelInterpretation "speakers" ["speakers", "discrete"]
  ~~SFInt32  [in,out] numberOfInputs        0          [0,∞)~~
  ~~SFInt32  [in,out] numberOfOutputs       0          [0,∞)~~
  # Mechanisms for parent-child input-output graph design remain under review
}
```

StreamAudioSource operates as an audio source whose media is received from a MediaStream obtained using the WebRTC or Media Capture and Streams APIs. This media source might originate from a microphone or sound-processing channed provided by a remote peer on a WebRTC call.

TODO confirm and describe attributes

16.4.21 WaveShaper

```
WaveShaper : X3DSoundProcessingNode {
  SFString [in,out] description           ""
  MFFloat  [in,out] curve                 []         [-1,-1]
  SFFloat  [in,out] gain                  1          [0,∞)
  SFString [in,out] oversample            "none"     ["none", "2x", "4x"]

  SFInt32  [~~in,~~out] channelCount          0          [0,∞)
  SFString [in,out] channelCountMode      "max"      ["max", "clamped-max", "explicit"]
  SFString [in,out] channelInterpretation "speakers" ["speakers", "discrete"]
  ~~SFInt32  [in,out] numberOfInputs        0          [0,∞)~~
  ~~SFInt32  [in,out] numberOfOutputs       0          [0,∞)~~
  # Mechanisms for parent-child input-output graph design remain under review
  MFNode   [in out] inputs                NULL       [X3DSoundAnalysisNode,X3DSoundChannelNode,X3DSoundProcessingNode,X3DSoundSourceNode]
}
```

WaveShaper represents a nonlinear distorter that applies a wave-shaping distortion curve to the signal. Non-linear waveshaping distortion is commonly used for both subtle non-linear warming, or more obvious distortion effects. Arbitrary non-linear shaping curves may be specified.

The *curve* field is an Array of floats numbers describing the distortion to apply.

The *oversample* field is specifies what type of oversampling (if any) should be used when applying the shaping curve. Allowed values follow. Note that for some applications, avoiding oversampling can produce a precise shaping curve.

- `"none"`: the curve is applied directly to the input samples with no oversampling.
- `"2x"`: oversample two times to improve the quality of the processing by avoiding some aliasing.
- `"4x"`: oversample four times for highest quality of the processing.

16.5 Support levels

The Sound component provides one level of support as specified in Table 16.2.

Table 16.2 — Sound component support levels

Level	Prerequisites	Nodes/Features	Support
1	Core 1 Time 1		
		X3DSoundSourceNode (abstract)	n/a
		X3DSoundNode (abstract)	n/a
		AudioClip	All fields fully supported.
		Sound	All fields fully supported.
2	Core 1 Time 1		
		All level 1 Sound nodes	All fields fully supported.

		X3DSoundAnalysisNode, *X3DSoundChannelNode*, *X3DSoundDestinationNode*, *X3DSoundProcessingNode*	All fields fully supported.
		Analyser, AudioBufferSource, AudioBufferSource, AudioDestination, BiquadFilter, ChannelMerger, ChannelSelector, ChannelSplitter, Convolver, Delay, DynamicsCompressor, Gain, ListenerPoint, MicrophoneSource, OscillatorSource, PeriodicWave, SpatialSound, StreamAudioDestination, StreamAudioSource, WaveShaper	All fields fully supported.