Technical Manual

OPTICODEC-PC 1010

AAC/HE AAC/aacPlus® Audio Streaming Encoder for Windows® 2000 / XP / 2003 Server

PRELIMINARY: Version 1.0 Software

IMPORTANT NOTE: Refer to the unit's rear panel for your Model #.

Model Number:	Description:
1010 LE	OPTICODEC-PC LE, AAC/HE AAC/aacPlus encoder for a single stream of 8-32 kbps, for use with any quality, Windows-compliant sound card.
1010 PE	OPTICODEC-PC PE, AAC/HE AAC/aacPlus encoder for multiple streams of 8-320 kbps, for use only with the Orban Optimod-PC sound card/audio processor.

MANUAL:

Part Number:	Description:
96127.100.01	OPTICODEC-PC Operating Manual

This symbol, wherever it appears, alerts you to important operating and maintenance instructions in the accompanying literature. Read the manual.

PLEASE READ BEFORE PROCEEDING!

Manual

Please review the Manual, especially the installation section, before installing the software.

Trial Period Precautions

If your unit has been provided on a trial basis:

You should observe the following precautions to avoid reconditioning charges in case you later wish to return the unit to your dealer:

Note the packing technique and save all packing materials. It is not wise to ship in other than the factory carton. (Replacements cost $35.00).

Packing

When you pack the software for shipping:

Wrap the unit in its original plastic bag to avoid abrading the paint.

If you are returning the unit permanently (for credit), be sure to enclose:

The Manual(s)
The Registration/Warranty Card
The installation CD

Your dealer may charge you for any missing items.

Trouble

If you have problems with installation or operation:

(1) Check everything you have done so far against the instructions in the Manual.

(2) Check the other sections of the Manual (consult the Table of Contents and Index) to see if there might be some suggestions regarding your problem.

(3) After reading the section on Factory Assistance, you may call Orban Customer Service for advice during normal California business hours. The number is +1 510 351-3500.

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OPTICODEC-PC 1010

AAC/HE AAC/aacPlus® Audio Streaming Encoder for Windows® 2000 / XP / 2003 Server

PRELIMINARY: Version 1.0 Software

aacPlus

by Coding Technologies

orban

OPTICODEC-PC is manufactured under license from Coding Technologies Inc.
and Dolby Laboratories Licesning Corporation.

This manual is part number 96127.100.01

orban

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Forward

Several years ago, one of the most successful radio stations in North America was CKLW in Windsor/Detroit. Innovative, technically superb audio complemented "The Big 8"'s excellent execution of their Top 40 program format to give the station a big signature sound that I have never forgotten. Its striving for perfection has been an inspiration and has influenced the goals, designs, and sounds of many Orban Optimod products to this day.

If it weren't for CKLW's innovative audio texture and the chances that it took to expose the world to many great artists, these discoveries would never have become part of the radio and music history that they now are. Just as many of these great artists have moved on from creating hit music to fusing their old styles with newer forms such as smooth jazz, Orban is evolving by giving the world new viable broadcast technology that build on its legacy.

I dedicate this technical manual to CKLW and those who made it the reality it once was, and to Leo, my faithful dog, who was beside me, night after night and day after day in the long process of preparing this document.

As the world moves on embracing innovative ways to deliver audio to an audience,
"ladies and gentlemen, the beat goes on…" – Bill Drake, radio programmer

– Greg J. Ogonowski

Table of Contents

Section 1

Introduction

About this Manual

The Adobe pdf form of this manual contains numerous hyperlinks and bookmarks. A reference to a numbered step or a page number (except in the Index) is a live hyperlink; click it to go immediately to that reference.

If the bookmarks are not visible, click the "Bookmarks" tab on the left side of the Acrobat Reader window.

This manual has a table of contents. To search for a specific word or phrase, you can use the Adobe Acrobat Reader's text search function.

Overview

Opticodec-PC is the first standards-based MPEG-4 AAC/aacPlus™, AAC/HE AAC, ISMA compliant and SHOUTcast/Icecast compatible encoding software for high quality streaming audio. Opticodec-PC offers the most important feature that the basic netcaster is looking for in an encoding product — entertainment-quality sound at economical bitrates.

The software lets streaming providers supply content encoded with the Coding Technologies® AAC/aacPlus codec, widely acknowledged as offering the highest available audio quality at the lowest possible bitrate. Streams encoded with Opticodec-PC can be experienced through RealPlayer® 10, QuickTime 6, Winamp 5.05, various Ethernet players, and 3G wireless devices. Streams can automatically list themselves on www.opticodec.net, a directory service for Opticodec-PC streams.

Opticodec-PC offers a choice of a standards-based RTSP/RTP streaming protocol for use with streaming servers (such as the free enterprise-class, scaleable Darwin Streaming Server from Apple) or the HTTP/ICY streaming protocol (for use with SHOUTcast or Icecast Servers). Both server types are non-proprietary and available for most computer platforms, and some servers are open-source.

Professional radio broadcasters would never consider going on the air without audio signal processing. They consider it a vital component of the program content, content being what attracts listeners. This carefully crafted content is what holds listeners and keeps them coming back. Broadcast ratings services have proven this true for over 30 years. Over that period, Orban's patented Optimod technology has helped radio and television broadcasters everywhere shape their sound to grab and hold their listening audiences.

Professional-grade netcasting requires audio processing similar to FM broadcast (although there are some important differences in the peak limiting because of the different characteristics of the pre-emphasized FM channel and the perceptually coded netcasting channel). Your listeners deserve to get the best quality and consistency you can provide. Good audio processing is one important thing that separates the amateur from the professional.

The Orban Optimod-PC 1100, a professional PCI sound card designed for streaming media, provides “genuine radio”™ audio processing for Internet broadcasters. With three on-board DSP's providing mixing, equalization, AGC, multi-band compression, and look-ahead limiting, Optimod-PC 1100, especially when combined with aacPlus encoding technology, delivers a polished and produced stream that has the same loudness, consistency, and punch as satellite and major-market FM radio. In addition to audio processing, Optimod-PC does internal and external audio mixing, leaving the CPU power available for encoding with Opticodec-PC. Together, Optimod-PC and Opticodec-PC provide a unique and tightly tuned system that offers the best audio quality streams possible with today’s technology.

Opticodec-PC is available in two versions, LE, and PE. Opticodec-PC LE, Light Edition, is compatible with all quality sound cards and encodes a single stream at bitrates between 8 and 32 kbps. Opticodec-PC PE, Professional Edition, is offered solely in a premium package coupled with an Optimod-PC and can encode multiple simultaneous streams at bitrates from 8 to 320 kbps; all streams carry the same Optimod-PC processed audio content. While the companion Optimod-PC will ordinary be used to process the stream for consistency and punch, it also comes with presets that allow it to do simple protection limiting.

Streaming Infrastructure Block Diagrams

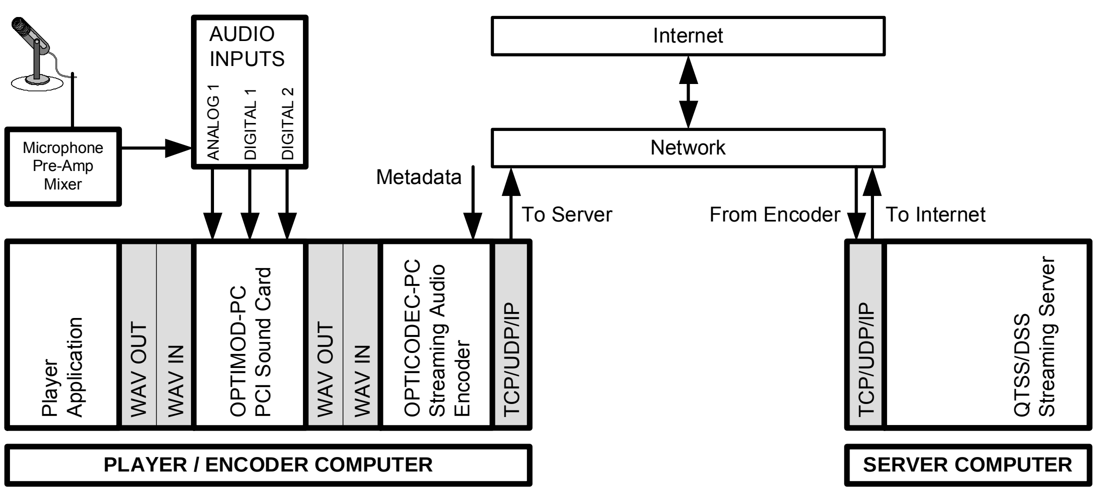


Figure 1-1: Typical streaming infrastructure where program material is sourced from a play-out system application with live assist

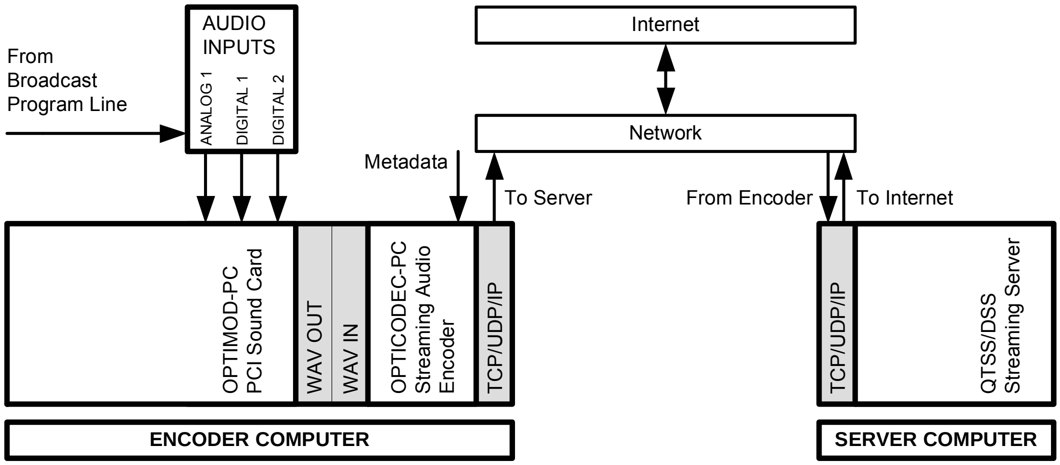


Figure 1-2: Typical streaming infrastructure where program material is sourced from a radio station on-air studio

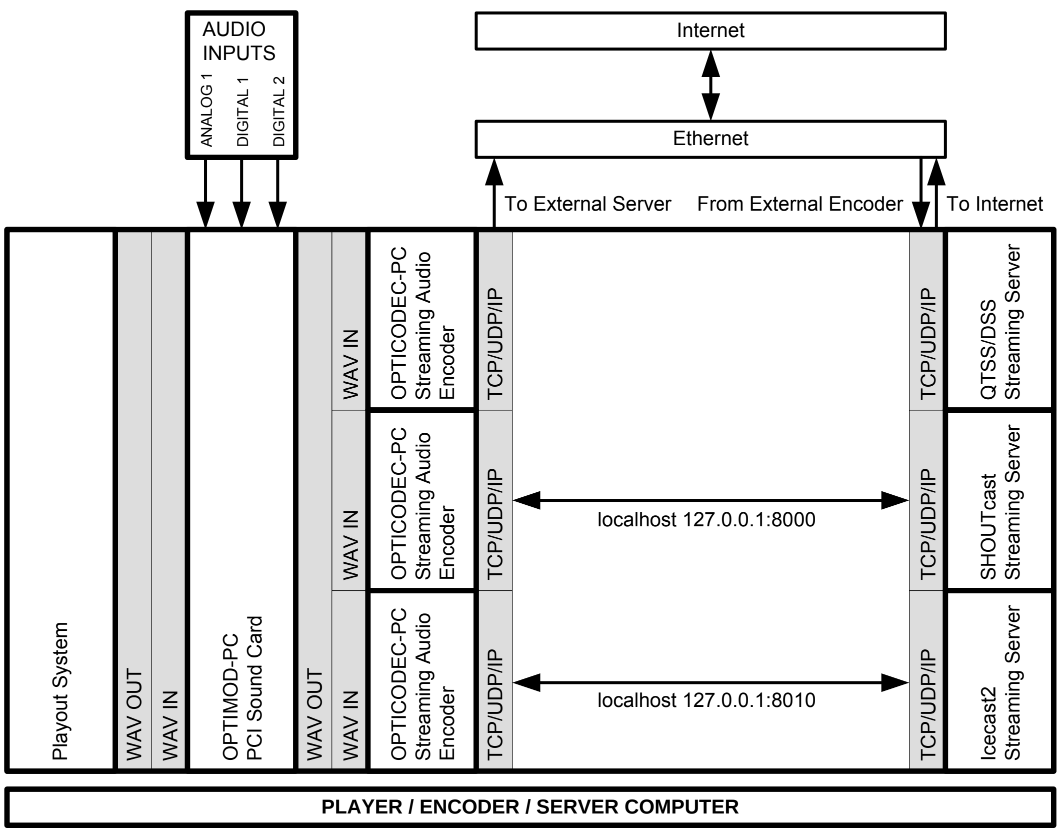


Figure 1-3: Typical multiple streaming encoder/server infrastructure where program material is sourced from a player application

Opticodec-PC offers the best available tradeoff between audio quality and bitrate. Compared to MP3, Opticodec-PC provides a better than 60% improvement in audio quality versus bitrate, reducing network streaming bandwidth requirements and costs accordingly. At 32 kbps, Opticodec-PC streams offer close to FM quality, without the phasey, watery character of other codecs operating at this bitrate. Many listeners prefer the audio quality of 48 kbps streams to FM.

There is a vast Internet and 3G wireless audience waiting for the entertainment-quality audio that Orban Opticodec-PC and Optimod-PC can provide.

Specifications and System Requirements

PE Version

COMPUTER

Minimum System Requirements:

Windows 2000:	Intel® Pentium II 400MHz, RAM = 64MB; 128MB recommended.
Windows XP:	Intel® Pentium II 400MHz, RAM = 128MB; 256MB recommended.
Windows 2003 Server	Intel® Pentium III 500 MHz RAM = 256MB; 512 MB recommended

This specification denotes the minimum CPU power necessary to control one OPTIMOD-PC card with external audio sources and one instance of the OPTICODEC-PC encoder. Additional cards, audio player and/or encoder software will require additional CPU power.

Processor and Chipset: This software has been tested and qualified with Intel CPUs and chipsets.

Sound Device: An Optimod-PC 1100 audio processor / sound card must be installed in the host computer in order to run the Opticodec-PC PE application.

Interface: Graphical User and Command-Line, batchable

ENCODER

Codec Technology: MPEG-2/MPEG-4 AAC/HE AAC /aacPlus v2 — Coding Technologies®

Sample Rates: 24 kHz, 32 kHz, 44.1 kHz, 48 kHz

Bitrates: 8, 10, 12, 16, 20, 24, 32, 40, 48, 56, 64, 80, 96, 128, 160, 192, 224, 256, 320 kbps

Number of Channels: 1-Mono / 2-Stereo

Coding Options: General; Voice

Number of Encoder Instances per Computer: Limited only by available CPU power.

STREAMING

Transport Protocols: RTSP/RTP, HTTP/ICY SHOUTcast, HTTP/ICY Icecast

RTP Payload Format: ISMA (audio/mpeg4-generic) / 3GPP/3GPP2 (audio/MP4A-LATM)

Method: **Unicast RTP/TCP** (External RTSP Server)

Transmission: Automatic Unicast – Announce – Session Description Protocol (.sdp) file per stream generated and transferred to server

Multicast RTP/UDP (Internal RTSP Server)

TTL: 255 default

Unicast HTTP/TCP

Packet Size: 1450 bytes plus IP Header Bytes = Total < 1500 byte MTU

Connection Fallback: Automatic Reconnection upon Connection Failure

Stream Information: Stream Name and Description; all server supported metadata

Metadata Input: Text File, Serial, Ethernet, Nullsoft Winamp

Server Requirements: Darwin Streaming Server 5.0 and later, QuickTime Streaming Server 5.0 and later, Nullsoft SHOUTcast DNAS 1.9.4 and later, Icecast2 2.0.2 and later

Server Platform: Available for Microsoft Windows 2000 Professional/Server, Windows 2003 Server, Windows XP Professional, Apple Mac OS X 10.2.8 and later Server and Proxy, Red Hat Linux 9, FreeBSD, Sun Solaris 9

LE Version

COMPUTER

Minimum System Requirements:

Windows 2000:	Intel® Pentium II 400MHz, RAM = 64MB; 128MB recommended.
Windows XP:	Intel® Pentium II 400MHz, RAM = 128MB; 256MB recommended.
Windows 2003 Server	Intel® Pentium III 500 MHz RAM = 256MB; 512 MB recommended

This specification denotes the minimum CPU power necessary to control one OPTIMOD-PC card with external audio sources and one instance of the OPTICODEC-PC encoder. Additional cards, audio player and/or encoder software will require additional CPU power.

Processor and Chipset: This software has been tested and qualified with Intel CPUs and chipsets.

Sound Device: Opticodec-PC LE will operate with any Windows-qualified sound card capable of the required sample rate and bit depth.

Interface: Graphical User and Command-Line, batchable

ENCODER

Codec Technology: MPEG-2/MPEG-4 AAC/HE AAC /aacPlus v2 — Coding Technologies®

Sample Rates: 24 kHz, 32 kHz, 44.1 kHz, 48 kHz

Bitrates: 8, 10, 12, 16, 20, 24, 32 kbps

Number of Channels: 1-Mono / 2-Stereo

Coding Options: General; Voice

Number of Encoder Instances per Computer: 1.

STREAMING

Transport Protocols: RTSP/RTP, HTTP/ICY SHOUTcast, HTTP/ICY Icecast2

RTP Payload Format: ISMA (audio/mpeg4-generic)

Stream Information: Name and Description

Method: **Unicast RTP/TCP** (External RTSP Server)

Transmission: Automatic Unicast – Announce – Session Description Protocol (.sdp) file per stream generated and transferred to server

Multicast RTP/UDP (Internal RTSP Server)

TTL: 255 default

Unicast HTTP/TCP

Packet Size: 1450 bytes plus IP Header Bytes = Total < 1500 byte MTU

Connection Fallback: Automatic Reconnection upon Connection Failure

Stream Information: Stream Name and Description, all server supported metadata

Metadata Input: Text File, Serial, Ethernet, Nullsoft Winamp

Server Requirements: Free Darwin Streaming Server 5.0 and later, QuickTime Streaming Server 5.0 and later, Nullsoft SHOUTcast DNAS 1.9.4 and later, Icecast2 2.0.2 and later

Server Platform: Available for Microsoft Windows 2000 Professional/Server, Windows 2003 Server, Windows XP Professional, Apple Mac OS X 10.2.8 and later Server and Proxy, Red Hat Linux 9, FreeBSD, Sun Solaris 9

These specifications are subject to design improvements and changes without notice.

Opticodec-PC TE, Test Edition is available upon request for testing encoder/server connectivity. With limited functionality, it allows testing network connectivity and authentication to verify server configuration.

Applications

Putting your audio content on the Internet or your LAN can be divided into three main steps: preprocessing the audio signal, encoding it, and streaming it to the network.

High quality streams begin with the cleanest possible audio source material. For best results, all material should be sourced in digital form to prevent any potential distortion from occurring in the analog-to-digital conversion process. CDs should be digitally extracted (ripped) to a PCM audio format if the digital storage system allows this, or to a 384 kbps or higher MPEG-1 Layer 2 format. Avoid Layer 3, as well as other codecs. More information on this topic can be obtained from the Orban publication, "Maintaining Audio Quality in the Broadcast Facility," available as a free download from http://www.orban.com.

Preprocessing

For optimum sound, loudness, and peak control, you should digitally preprocess the Internet audio signal to condition it prior to encoding. The appropriate preprocessing has much in common with the preprocessing required for DAB, HD Radio™, CD mastering, or digital satellite.

Preprocessing is necessary for several reasons. Automatic gain control and equalization achieve a consistent sound, while accurate peak control maximizes loudness.

Preprocessing each program element before it is stored on a playout system is not as effective as preprocessing the mixed audio on the program line immediately before it is streamed. The latter technique maximizes the smoothness of transition between program elements and makes voice from, announcers, or presenters merge smoothly into the program flow, even if the announcer is talking over music.

Peak clipping sounds terrible in digital systems because these systems do not rely on pre-emphasis/de-emphasis to reduce audible distortion. Instead of peak clipping, the best sounding processors use some form of look-ahead limiting. The carefully peak limited signal is then digitally connected to Opticodec-PC to preserve the audio signal waveform integrity.

Orban Optimod-PC (recommended for Opticodec PC LE and required for Opticodec-PC PE to operate) is a PCI sound card with on-board digital signal processing that is suitable for both live streaming and on-demand programming. Its three on-board Motorola DSP56362 DSP chips provide a loud, consistent sound to the consumer by performing automatic gain control, equalization, multiband gain control, and peak-level control. Optimod-PC's sound card emulation allows it to talk through the operating system via the Windows' WAVE mechanism to Opticodec-PC, running on the same computer that houses Optimod-PC.

While there are several types of audio processors available other than Optimod-PC, conventional AM, FM, or TV audio processors that employ pre-emphasis/de-emphasis and/or clipping peak limiters are most inappropriate for use with perceptual audio coders such as Opticodec-PC. The pre-emphasis/de-emphasis limiting in these devices unnecessarily limits high frequency headroom. Further, their clipping limiters create high frequency components— distortion—that the perceptual audio coders would otherwise not encode. None of these devices has the full set of audio and control features found in Optimod-PC.

Without Optimod-PC processing, audio can sound dull, thin, or inconsistent in any combination. Optimod-PC's multiband processing automatically levels and re-equalizes its input to the "major-market" standards expected by the mass audience. Broadcasters have known for decades that this polished, produced sound attracts and holds listeners.

You can expect a very large increase in loudness from Optimod-PC processing by comparison to unprocessed audio (except for audio from recently mastered CDs, which are often overprocessed in mastering). Broadcasters generally believe that loudness relative to other stations attracts an audience that perceives the station as being more powerful than its competition. We expect that the same subliminal psychology will hold in netcasting too.

Remote Access & Control:

Optimod-PC has the unique ability to be remotely accessed and controlled over any TCP/IP network. After the appropriate security and administration setup, Optimod-PCs I/O mixer, processing parameters, and presets can be controlled from anywhere, including from other applications.

Mixing Facilities:

In addition to sound card and audio processing functionality, Optimod-PC is also a capable mixer, having one stereo analog input, two AES3 / SPDIF digital inputs (which can accept any sample rate from 32 to 96 kHz), and one WAVE input (to accept Windows sound sources), all of which can be mixed. Thanks to onboard sample rate converters, the two digital inputs can accept and mix asynchronous sources, which may have different sample rates. In practice, the four inputs might be used for a local feed, a network feed, a voice channel, and a wave player, making Optimod-PC the heart of a "desktop netcasting studio." In many cases, this versatility allows you to avoid use of an external mixing desk, thereby keeping the audio path 100% digital. The wave player could be any one of a number of broadcast-oriented automated playout systems.

Using Optimod-PC's separate "processed" and "unprocessed" mixers, any of the inputs in any combination can be processed or passed directly to the input of Opticodec-PC without processing—you can always choose how much processing (if any) to apply to the audio. These features allow local program insertion, such as those required in order to address the broadcast rights issues of many commercials, programs, and events.

Because it uses Microsoft DirectSound Drivers, Optimod-PC is able to play multiple audio streams from multiple audio sources, eliminating the need for multiple or multi-channel sound devices for professional playout systems used in automation mode. Given the CPUs available today, MPEG1 Layer 2, and/or Layer 3 decoding can occur at the operating system level, eliminating the requirement for expensive hardware-based MPEG decoder sound devices.

Encoding

Opticodec-PC receives the output of Optimod-PC, which looks like a sound card to the operating system. Opticodec-PC then reduces the bitrate of the processed signal by applying it to an AAC or aacPlus perceptual coder and packetizing the resulting data for an Ethernet network. When the encoder connects to the streaming server, the encoder generates the Session Description Protocol file and transfers it automatically to the streaming server.

The most basic use of Opticodec-PC is to create a single stream at a single bitrate. However, the output of a given Optimod-PC card can feed several Opticodec-PC encoders running at different bitrates to service different audience bandwidths; all of these streams will carry the same audio program.

If you need more than one audio program stream, use multiple Optimod-PC cards (some of which can be housed in one or more PCI expansion chassis). If you need multiple streams at different bitrates, configure each Optimod-PC card to feed its own array of Opticodec-PC PE encoders.

Each installation of Opticodec-PC PE is keyed to one Optimod-PC card, so running more than one audio program stream requires one Opticodec-PC PE installation per audio program stream even if all of these installations are on one computer. However, a single Opticodec-PC installation can create multiple streams at different bitrates if all of these streams contain the same audio program.

About Perceptual Coders

CD-quality audio (16-bit words at 44.1 kHz sample rate) requires 705,600 bits per second per channel, which is far too high for economical streaming. Perceptual coding reduces the number of bits per second necessary to transmit a high-quality audio signal.

Perceptual coders exploit models of how humans perceive sound. In particular, perceptual coders exploit the phenomenon of *psychoacoustic masking*. This means that louder sounds will "drown out" (or "mask") weaker sounds occurring at the same time, particularly if the frequency of the louder sound is close to the weaker sound's frequency. Loud sounds not only mask weak sounds occurring simultaneously in time (spectral masking), but can also drown out weak sounds occurring a few milliseconds before the loud sound starts or a few milliseconds after it stops (temporal masking).

The basic principle of perceptual coding is to divide the audio into frequency bands and then to code each frequency band with the minimum number of bits that will yield no audible change in that band. Reducing the number of bits used to encode a given frequency band raises the quantization noise floor in that band. If the noise floor is raised too far, it can become audible and cause artifacts.

A second major source of artifacts in codecs is pre- and post-echo caused by ringing of the narrow bandpass filters used to divide the signal into frequency bands. This ringing worsens as the number of bands increases, so some codecs may adaptively switch the number of bands in use, depending on whether the sound has significant transient content. This ringing manifests itself as a smearing of sharp transient sounds in music, such as those produced by claves and wood blocks.

Psychoacoustic Models

Perceptual coders exploit complex models of the human auditory system to estimate whether a given amount of added noise can be heard. They then adjust the number of bits used to code each frequency band such that the added noise is undetectable by the ear if the total "bit budget" is sufficiently high. Because the psychoacoustic model in a perceptual coder is an approximation that never exactly matches the behavior of the ear, it is desirable to leave some safety factor when choosing the number of bits to use for each frequency band. This safety factor is often called the "mask-to-noise ratio," measured in dB. For example, a mask-to-noise ratio of 12 dB in a given band would mean that the quantization noise in that band could be raised by 12 dB before it would be heard. (That is, there is a safety margin of two bits in that band's coding.) For the most efficient coding, the mask-to-noise ratio should be the same in all bands, ensuring that the sound elements equitably share the available bits in the transmission channel.

Increasing the number of bits per second in the transmission always improves the mask-to-noise ratio. It is important to allocate extra bits to the transmission if the audio will be processed after it has been decoded at the output of the perceptual coder (for example, by a second "cascaded" perceptual coder, or by a multiband audio processor such as Optimod-PC). Done correctly, this increased bitrate will raise the mask-to-noise ratio far enough to prevent downstream processing from causing the noise to become unmasked.

Because it occurs in narrow frequency bands, unmasked noise does not sound like familiar white noise at all. Instead, it most often sounds like distortion, or like warbling, comb filtering, or gurgling—an "underwater" sound.

Coding Efficiency

Different sounds will vary greatly in the efficiency with which a perceptual coding system can encode them. Therefore, for a constant transmission bitrate, the mask-to-noise ratio will constantly change. Pure sounds having an extended harmonic structure (such as a pitch pipe) are particularly difficult to encode because each harmonic must be encoded, the harmonics occupy many different frequency bands, and the overall spectrum has many "holes" that are not well-masked, so that added noise can be easily heard. The output of a multiband audio processor that uses clipping is another sound that is difficult to encode, because the clipper creates added distortion spectrum that does not mask quantization noise well, yet may cause the encoder to waste bits when trying to encode the distortion.

Sophisticated encoders use a short "bit reservoir" to save up unused bits so they can be applied to difficult-to-encode sounds. However, the length of the bit reservoir will directly affect the coding delay, so dynamic allocation of bits occurs only over rather short time windows (in the order of tens of milliseconds). Another feature of sophisticated encoders is "redundancy reduction," which encodes frequently appearing data with shorter digital words and infrequently appearing data with longer words.

Encoding Stereo

Usually, there is some correlation between the left and right channels of a stereo signal. At lower bitrates, one way to achieve higher quality is to exploit this correlation when coding stereo information:

Depending on program content, the encoder dynamically switches between discrete left/right coding and sum-and-difference coding. The difference signal often requires fewer bits than the sum signal to encode with high audible quality, thereby saving bits in the overall coding of the stereo signal.

There is no benefit to joint stereo coding when the two channels contain independent information because there is no correlation between the channels.

Opticodec-PC Codecs

Opticodec-PC offers two coding algorithms from the several standardized by ISO/MPEG (Moving Pictures Experts Group): the AAC and aacPlus® v2 algorithms.

AAC is intended for very high quality coding with compression up to 12:1. The AAC codec is about 30% more efficient than MPEG1 Layer 3 and about twice as efficient as MPEG1 Layer 2. The AAC codec can achieve "transparency" (that is, listeners cannot audibly distinguish the codec's output from its input in a statistically significant way) at a stereo bitrate of 128 kb/sec, while the Layer 2 codec requires about 256 kb/sec for the same quality. The Layer 3 codec cannot achieve transparency at any bitrate, although its performance at 192 kbps and higher is still very good.

AAC stands for Advanced Audio Coding. Intended to replace Layer 3, AAC was developed by the MPEG group that includes Dolby, Fraunhofer (FhG), AT&T, Sony, and Nokia—companies that have also been involved in the development of audio codecs such as MP3 and AC3 (also known as Dolby Digital™).

AAC does not stand for Apple Audio Codec, although Apple was one of the first to implement this technology with the introduction of Apple iTunes and QuickTime 6.

The Coding Technologies “Spectral Band Replication” (SBR) process can be added to almost any codec. This system transmits only lower frequencies (for example, below 8 kHz) via the codec. The decoder at the receiver creates higher frequencies from the lower frequencies by a process similar to that used by “psychoacoustic exciters.”

Channel Mode	Bitrate [kbps]	Sample rate [kHz]	Audio Bandwidth [kHz]
1ch-Mono	8	24	8.3
1ch-Mono	10	24, 32	10.9, 11.0
1ch-Mono	12	24, 32, 44.1	11.4, 11.0, 10.7
1ch-Mono	16	32, 44.1, 48	12.3, 11.7, 12.0
1ch-Mono	20	32, 44.1, 48	14.5, 14.8, 15.4
1ch-Mono	24	32, 44.1, 48	15.3, 14.8, 15.4
1ch-Mono	28	32, 44.1, 48	16.0, 16.2, 16.9
1ch-Mono	32	32, 44.1, 48	16.0, 16.2, 16.9
1ch-Mono	40	32, 44.1, 48	16.0, 17.6, 18.4
1ch-Mono	48	32, 44.1, 48	16.0, 20.3, 20.3
1ch-Mono	56	32, 44.1, 48	16.0, 20.3, 20.3
1ch-Mono	64	32, 44.1, 48	16.0, 20.3, 20.3
2ch-Stereo	24	32, 44.1, 48	13.8, 12.7, 13.1
2ch-Stereo	28	32, 44.1, 48	15.3, 14.8, 15.4
2ch-Stereo	32	32, 44.1, 48	15.3, 14.8, 15.4
2ch-Stereo	40	32, 44.1, 48	16.0, 16.2, 16.9
2ch-Stereo	48	32, 44.1, 48	16.0, 16.2, 16.9
2ch-Stereo	56	32, 44.1, 48	16.0, 17.6, 18.4
2ch-Stereo	64	32, 44.1, 48	16.0, 20.3, 22.1
2ch-Stereo	80	32, 44.1, 48	16.0, 20.3, 22.1
2ch-Stereo	96	32, 44.1, 48	16.0, 20.3, 22.1
2ch-Stereo	112	32, 44.1, 48	16.0, 20.3, 22.1
2ch-Stereo	128	32, 44.1, 48	16.0, 20.3, 22.1

Table 1-1: aacPlus Audio Bandwidth vs. Bitrate, Sample rate, and Channel Mode

Channel Mode	Bitrate [kbps]	Sample rate [kHz]	Audio Bandwidth [kHz]
1ch-Mono	16	24	5.2
1ch-Mono	20	24	7.2
1ch-Mono	24	24, 32	7.2, 7.2
1ch-Mono	28	24, 32	10.0, 10.0
1ch-Mono	32	24, 32, 44.1, 48	10.0, 10.0, 10.0, 10.0
1ch-Mono	40	24, 32, 44.1, 48	12.0, 12.4, 12.4, 12.4
1ch-Mono	48	24, 32, 44.1, 48	12.0, 13.5, 13.5, 13.5
1ch-Mono	56	24, 32, 44.1, 48	12.0, 15.5, 15.5, 15.5
1ch-Mono	64	32, 44.1, 48	15.5, 15.5, 15.5
1ch-Mono	80	32, 44.1, 48	16.0, 17.6, 17.6
1ch-Mono	96	32, 44.1, 48	16.0, 22.1, 24.0
1ch-Mono	112	32, 44.1, 48	16.0, 22.1, 24.0
1ch-Mono	128	32, 44.1, 48	16.0, 22.1, 24.0
1ch-Mono	160	32, 44.1, 48	16.0, 22.1, 24.0
2ch-Stereo	24	24	6.6
2ch-Stereo	28	24	6.6
2ch-Stereo	32	24	6.6
2ch-Stereo	40	24, 32	8.5, 8.5
2ch-Stereo	48	24, 32	8.5, 8.5, 8.5
2ch-Stereo	56	24, 32, 44.1, 48	12.0, 12.5, 12.5, 12.5
2ch-Stereo	64	32, 44.1, 48	12.5, 12.5, 12.5
2ch-Stereo	80	32, 44.1, 48	13.5, 13.5, 13.5
2ch-Stereo	96	32, 44.1, 48	15.5, 15.5, 15.5
2ch-Stereo	112	32, 44.1, 48	16.0, 16.0, 16.0
2ch-Stereo	128	32, 44.1, 48	16.0, 16.0, 16.0
2ch-Stereo	160	32, 44.1, 48	16.0, 17.6, 17.6
2ch-Stereo	192	32, 44.1, 48	16.0, 22.1, 24.0
2ch-Stereo	224	32, 44.1, 48	16.0, 22.1, 24.0
2ch-Stereo	256	32, 44.1, 48	16.0, 22.1, 24.0
2ch-Stereo	320	32, 44.1, 48	16.0, 22.1, 24.0

Table 1-2: AAC Audio Bandwidth vs. Bitrate, Sample rate, and Channel Mode

A low-bandwidth signal in the compressed bit stream provides “hints” to modulate these created high frequencies so that they will match the original high frequencies as closely as possible. Adding SBR to the basic AAC codec creates aacPlus, which offers the best subjective quality currently available at bitrates below 128 kbps. At bitrates below 128 kbps, full subjective transparency cannot be achieved at the current state of the art, yet the sound can still be very satisfying. (In the phraseology of the ITU 1 to 5 subjective quality scale, this means that audible differences introduced by the codec are judged by expert listeners to be “detectable, but not annoying.”)

Coding Technologies’ aacPlus v2, the latest in MPEG-4 Audio and previously known as "Enhanced aacPlus," is aacPlus coupled with the new MPEG Parametric Stereo technique created by Coding Technologies and Philips. Where SBR enables audio codecs to deliver the same quality at half the bitrate, Parametric Stereo enhances the codec efficiency a second time for low-bitrate stereo signals. Both SBR and Parametric Stereo are backward- and forward-compatible methods to enhance the efficiency of any audio codec. As a result, aacPlus v2 delivers streaming and downloadable 5.1 multichannel audio at 128 Kbps, near CD-quality stereo at 32 Kbps, excellent quality stereo at 24 Kbps, and great quality for mixed content down to 16 Kbps and below.

MPEG standardized Coding Technologies’ aacPlus as MPEG-4 HE AAC (MPEG ISO/IEC 14496-3:2001/AMD-1: Bandwidth Extension). With the addition of MPEG Parametric Stereo (MPEG ISO/IEC 14496-3:2001/AMD-2: Parametric coding for high quality audio), aacPlus v2 is the state-of-the-art in low bitrate open standards audio codecs. The Coding Technologies codecs provide the absolute best possible sound per bit the current state-of-the-art will allow, without the typical resonant, phasey, watery character of other codecs.

Trading-Off Audio Bandwidth against Bitrate, Sample rate, and Channel Mode

High audio bandwidth does not guarantee good sound in codecs. In many cases, especially at low bitrates, it is actually just the opposite. For example, FM radio is a 15 kHz medium, yet there are plenty of codecs claiming to have 20 kHz response that sound much worse than FM radio.

The designers of the various codecs usually determine the optimum tradeoff between bitrate, sample rate, and channel mode (stereo or mono) by performing extensive listening tests. To maximize overall audio quality at lower bitrates, it is important to allocate the bits efficiently. This usually means allocating more bits to those frequency ranges most important to music and speech.

Below a certain sample rate (which depends on the design of the individual codec), codec designers have determined that limiting audio bandwidth to less than 20 kHz achieves highest overall quality. For example, AAC requires 192 Kbps or more for 20 kHz+ response (*Table 1-2* on page 1-12) and aacPlus requires 64 Kbps or more for 20 kHz+ response (*Table 1-1* on page 1-11).

We recommend using aacPlus v2 for stereo streams below 48kbps. Be sure your target players support it; otherwise, the streams will play in mono.

Cascading Codecs

There are two general applications for codecs in broadcasting — "contribution" and "transmission." A contribution-class codec is used in production. Accordingly, it must have high enough "mask to noise ratio" (that is, the headroom between the actual codec-induced noise level and the just-audible noise level) to allow its output to be processed and/or to be cascaded with other codecs without causing the codec-induced noise to become unmasked. A transmission-class codec, on the other hand, is the final codec used before the listener's receiver. Its main design goal is maximum bandwidth efficiency. Some codecs, like Layer 2, have been used for both applications at different bitrates (and Layer 2 continues to be used as the transmission codec in the Eureka-147 DAR system and many DBS satellite systems). However, assuming use of an MPEG codec, modern practice is to use Layer 2 for contribution only (minimally at 256 kbps, with 384 kbps preferred), reserving transmission for AAC or aacPlus. Layer 3 has become a consumer format, and even that is being replaced by the next generation AAC/HE AAC/aacPlus.

The most general operational advice is this:

- **Use compression only when necessary**. Hard drives have become very inexpensive, and there is little excuse for excessively compressing a source library. Linear PCM is best.
- A good codec such as AAC/HE AAC/aacPlus requires a good source to produce excellent results. If you must use **compression in production or transmission** ahead of the audio preprocessor (like Optimod-PC) and have the luxury of high bitrates, **use Layer 2 at 128 kb/sec/channel or above (256 kb/sec stereo)**. This will be audibly transparent for as many as ten passes. Avoid Layer 3 sources; Layer 3 was never rated transparent at any bitrate.
- **Do not use a higher sampling frequency than necessary**. 32 kHz is adequate for AM, FM. analog television, and low bitrate (~32 kbps) streaming. However, if you are creating a hard-disk music library and plan to use it for DAB or high bitrate streaming now or in the future, 44.1 kHz will yield CD-quality bandwidth 20 kHz frequency response. Especially with low bitrate codecs, a 32 kHz sample rate is generally optimum and sounds better than higher sample frequencies because the bit allocation for the codec is concentrated in the most audible region of the audio spectrum. This is a case where less is truly more.
- **Carefully monitor any cascade of codecs by listening tests**. There are an infinite number of combinations possible, and the human ear must be the final arbiter of quality. Be particularly sensitive to loss of "snap" and transient definition, loss of stereo imaging, loss of very high frequencies, comb-filtering or "underwater" sounds, and buildup of distortion.
- **Do not use Microsoft Windows Media Player to play MPEG-1 Layer 2 files.** There is a confirmed problem with the MPEG-1 Layer 2 decoder filter used in the current and several past releases of Windows Media Player. This filter causes a poor signal-to-noise ratio in the form of low-level noise that is only there when the least significant bits are present. It is audible during quiet portions of audio and prevents the filter from being usable in professional applications. Audio signal processing will make this more apparent. We hope that Microsoft will someday address this issue.

Networking

Opticodec-PC supports both unicast and multicast streams. Each method has its own advantages and your streaming application will determine which one to use.

To connect to the Internet using unicast, a server is required. This receives the output of the encoder and creates the streams to which your listeners connect. Opticodec-PC supplies an output compatible with the free Darwin Streaming Server, which is available for multiple platforms including Linux®, FreeBSD®, Sun Solaris®, Microsoft Windows®, and QuickTime Streaming Server for Apple Macintosh®. It is also compatible with the SHOUTcast DNAS and the Icecast2 servers, also freely downloadable.

Network Bandwidth Considerations

If you have access to large bandwidth Internet connectivity, you could conceivably run the server software on your encoder computer—just connect the computer to an Ethernet Internet feed and you are ready to go. However, most netcasters do not have that option because the studio or program origination is in one place and the Internet service provider (ISP) is somewhere else. If that's the case, the best and most economical way to connect is to establish what's called a "co-lo," or co-location, which requires running your own server software on another computer, locating that computer at the ISP, and running one stream per program from your encoder to the server. Typically, this requires a full-time, non-dial-up dedicated connection from your encoder to your ISP. Bandwidth requirements for this connection depend upon the bitrate and number of streams being sent to the server.

A high reliability connection is also recommended to prevent encoder-server disconnects, although Opticodec-PC has the ability to automatically reconnect when this occurs. If reliability is the goal, avoid consumer Internet connections, especially cable Internet and some DSL. The relatively small upload bandwidth available from consumer Internet services will severely limit the encoder and/or server. The reliability of these services is generally not good enough for continuous streaming. Furthermore, running a server on this type of Internet service may break your Internet service agreement.

Many ISPs provide servers and administration services to run the appropriate streaming server software. Although you are not responsible for the server administration in this scenario, it comes at a price.

We have just described how to get your program on the network. Here is where the listeners come in. There are different ways that people can connect to your stream.

- Most Internet streams are implemented via **unicasting,** which requires a single, independent connection to the server for each stream. (See *Unicast* on page 1-24.)
- In a **multicast**, a single stream is shared among the player clients. Although this technique reduces network congestion, it requires a network that either has access to the multicast backbone (otherwise called the *Mbone*) for content generally distributed over the Internet, or is multicast-enabled for content distributed within a contained private network. Multicast streams are sent directly to a group address, such an IP multicast address, which many client com-

puters can simultaneously access. The users of a multicast have no control over the media content. Multicasts are an efficient way to deliver the same material to a group of people over a LAN, as only one copy of the stream is sent over the network. (See *Multicast* on page 1-25.)

Because Opticodec-PC contains a multicast server, more than one listener can connect to the same IP address without increasing network traffic. This is an excellent way to deliver corporate or academic content to an internal audience or to stream radio stations to the staff at their computer workstations. Unless your LAN contains a router that is not multicast enabled and that separates the encoder from your listeners, you do not need to use a server to multicast within a LAN. For listeners to connect to your stream via a typical LAN, they have to connect their decoder applications to the same local IP address as the one you assigned to the output of Opticodec-PC.

Bandwidth Requirements

Streaming puts demand on your server system in a number of ways, the most important being bandwidth. For example, three different unicast streams for different purposes will attract different audiences with different network connectivity requirements.

Stream Type	Attendance	Audience Connection Speed	Total Concurrent Bandwidth + 20%	Total Throughput for 1 Hour
Distance Learning	100	20 kbps	2.4 Mbps	990 MB
Small Corporate Meeting	100	32 kbps	3.8 Mbps	17.1 MB
Medium Entertainment Stream	1000	48 kbps	57.6 Mbps	25.9 GB
Large Entertainment Stream	5000	48 kbps	288 Mbps	129.6 GB

Table 1-3: Bandwidth Requirements for Typical Network Streams

Even the smallest academic streams can generate huge numbers that require more than a single E-1 or T-1 line to serve. Corporate and entertainment streams can sometimes require multiple E-3 or T-3 lines, or even higher capacity to serve. Large streams may even require more than one server to handle the necessary network throughput. However, since Orban Opticodec-PC is bandwidth efficient, you are able to serve a larger audience at a lower cost of operation with higher audio quality than with inferior older generation codecs.

Not all networks have 100% of their theoretical capacity available for data transfer. You are practically limited to about 80% of theoretical maximum because of the way TCP/IP traffic is handled on a network. For example, a 100 Mbps LAN is limited to about 80 Mbps. In addition to this practical limitation, you may want to allot additional bandwidth for other tasks, such as file transfers and backup procedures. An additional 10% should suffice. Generally, a good equation for calculating the practical capacity of a network is:

```
Practical Network Capacity = Theoretical Maximum * 70%
Maximum Simultaneous Streams = Practical Network Capacity / Stream Bitrate
```

For example, if you have a 100 Mbps network and you want to know how many 32 kbps streams you can support, the equation is 10,000,000 * 70% / 32000 = 2187 simultaneous streams.

Your required network bandwidth will be determined by your intended audience size and whether you are using unicast or multicast streaming.

- Unicast streaming bandwidth, the most common, is the total number of simultaneous streams multiplied by the stream bitrate, plus some network overhead, or:

```
Bandwidth (Unicast) = (Number of Concurrent Streams) * (Stream Bitrate) + 20%
```

- Multicast streaming bandwidth is simply that of a single stream plus some network overhead, or:

```
Bandwidth (Multicast) = (Stream Bitrate) + 20%
```

To ensure reliability, bandwidth projections must be based on peak usage, not average usage.

Streaming Architecture

Live Streaming

Live events, such as radio broadcasts, concerts, speeches, lectures, and sporting events are commonly streamed over the Internet as they happen with the assistance of broadcasting encoding software such as Orban Opticodec-PC Streaming Encoder. The broadcasting software encodes a live source, such as studio originated audio, in real time, and delivers the resulting stream to the server. The server then serves, reflects, and/or relays the live stream to media player clients.

Like a radio broadcast, a live stream provides identical, essentially synchronized content to all listeners. (The only factor that prevents perfect synchronization is varying network latencies between the server and the various listeners.) This live experience can be simulated with recorded content by broadcasting from an archive source such as a playout system or by creating playlists of media on a media server.

File Streaming

With file streaming, or on-demand delivery, such as archived broadcasts, concerts, speeches, lectures, and sporting events, each user initiates the stream from the beginning, so no user ever comes in late to the stream. No broadcasting or streaming encoder software is required. The files are encoded prior to upload to the server using software such as Orban Opticodec-PC File Encoder.

Overview of Streaming Architecture

There are several ways to stream content over a network. This can be confusing at first, because some of these ways are only slightly different from others. The terminology associated with them and the names that commonly refer to them can be equally confusing. The following explanation should you to decide which server platform or platforms are best for your streaming application.

To stream content, a streaming server is commonly used to control and deliver live and/or file streams, just as a web server delivers web pages and files. However, there are exceptions. Furthermore, there are important differences in the ways that different servers function.

Web servers cannot deliver live streams. Although web servers can allow a client player to play a file, playback begins when the player's buffer fills, which occurs sometime after the web server starts sending the file to the user's computer. This type of stream is called a *progressive download* or a *fast start stream*. There is no content protection because the determined user can always find a way to download your content even if this has not happened automatically, which it usually does.

Conversely, the content bitrate determines how quickly a streaming server sends media content to a player client. Streaming servers do not send files to the user's hard drive, thereby protecting the content.

Each server type has its own features and is compatible with its own set of client players. To serve a greater audience, you might consider using more than one server platform.

Network Transports

Different servers use different network transports and protocols. Knowing the structure of the transports and protocols used for the different server technologies will help you understand the differences in these servers. *Figure 1-4* shows the various

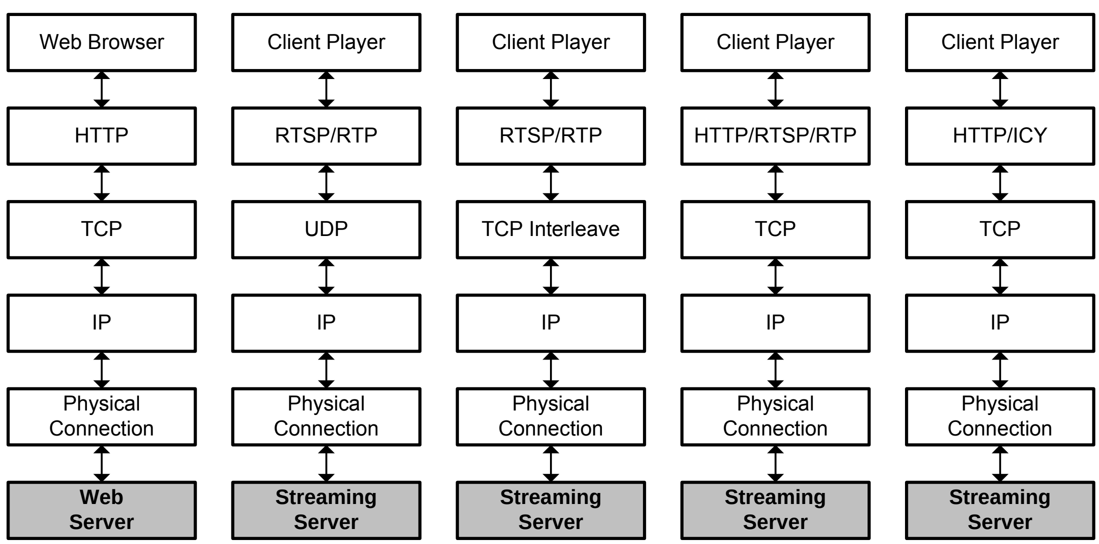


Figure 1-4: Server Transports and Protocols

network transports and protocols used with their associated servers.

Streaming Server Advantages

Streaming servers open a dialog with the media player. There are two sides to this dialog — one for passing control messages between the media player client and the server and one for transferring the media content. Because they continue to exchange control messages with the player, streaming servers can adjust to changing network conditions as the content plays, improving the user experience. The control messages also include user commands like "play," "pause," "stop," and "seeking to a particular part of the file."

Compared to posting downloadable media files to your website, there are many advantages to streaming:

- Users can stream your media immediately. There are no lengthy downloads, regardless of network connectivity.
- Streaming is the only way to distribute live media content such as radio stations, playlist streams, entertainment, news, and sporting events.
- Streaming media files are not limited to file sizes that make a reasonable download. Long-form media content (such as complete programs, concerts, and archives) would make multi-megabyte downloads, yet stream effortlessly.
- Users have complete control over the streaming media content by being able to pause, seek, and play only the parts they want.
- Depending upon the server platform in use, multicasting can allow many users to connect into one stream, dramatically saving bandwidth.
- Streaming allows you to maintain control over the distribution and copyright of your media. Anyone can download media content, alter it, and redistribute it. Although not impossible, it is much harder to redistribute the contents of a stream.

Web Server Advantages

It is important to understand the difference between HTTP used on a web server and HTTP/ICY used on a streaming server, and not to confuse the two. HTTP is used to deliver web pages to a browser and/or to download files. HTTP/ICY is used to deliver streaming media content to a client media player. They both use TCP/IP as their main data transport.

By using web servers, HTTP can deliver fast start or progressive download streaming, which is not really streaming at all. It is a file download and starts sending data as fast as the network connection will allow. To the user, it is similar to streaming that has no stream control features. You cannot move to the end of the file until the entire file has downloaded.

There are some advantages to making your media content available for download:

- It is easy to implement. Encode your files and put them on your web server with the appropriate HTML links authored to standard web pages.

- No special server software is required. Your web server software can deliver media content.
- Files downloaded to the user's computer can be used later without additional network connectivity.
- Media content downloads to the user's computer as fast as the network connection will allow.
- Media content gets to the user no matter how slow the network connection.
- With fast network connections, media content can play as it downloads. However, there are no advanced stream control facilities.
- Lost packets are retransmitted until they are received.
- There are no problems with routers and/or firewalls.

RTSP/RTP — Streaming Servers

RTSP/RTP streaming servers can use HTTP and TCP/IP to deliver media content streams, but by default, they use protocols such as RTSP and UDP/IP. RTSP provides built-in support for the control messages and other features of streaming servers.

UDP is a lightweight protocol that saves bandwidth by introducing less overhead than TCP/IP. It emphasizes continuous delivery instead of being 100% accurate, a feature that makes it well suited to real-time operations like streaming. Unlike TCP, it does not request resends when packets are missing. With UDP, if a packet gets dropped on the way from the server to the client player, the server just keeps on sending data. UDP's philosophy is that it is better to have a momentary glitch in the media content than to stop everything and wait for the missing data to arrive.

UDP has the following properties:

- It can be used for live and file streaming, both unicast and/or multicast.
- It requires a streaming server and/or live streaming encoder.
- It never uses more bandwidth than it needs. Only the data for the part of the file that is used is transferred.
- Real-time streaming allows the user to view long streams or continuous transmissions without having to store more than a few seconds of data locally, which is then discarded.
- It does not leave a copy of the media content on the user's computer.
- Using RTP transmission under RTSP control, a user can seek to any point in a file or clip on a streaming server without downloading the file.
- The stream will drop out if data rate exceeds network connection speed.
- RTP uses UDP/IP protocol by default, which doesn't attempt to retransmit lost packets. UDP is faster and more efficient than TCP/IP but lacks a mechanism for reporting lost packets, so streaming over congested networks almost always results in some data loss in the form of dropouts. UDP/IP allows multicasts as well as live streams, which are both cases where retransmission might not be practical.

- Optional Reliable UDP, if supported by the client player, can be used for a higher level of retransmission of lost packet control than TCP/IP.
- Firewalls and/or routers can stop UDP/IP.
- RTP can be interleaved with the RTSP control connection, which uses TCP/IP. This can get through firewalls that block UDP/IP. However, form of RTP is less efficient and adds server load, a slight network overhead, and increased bi-trate.

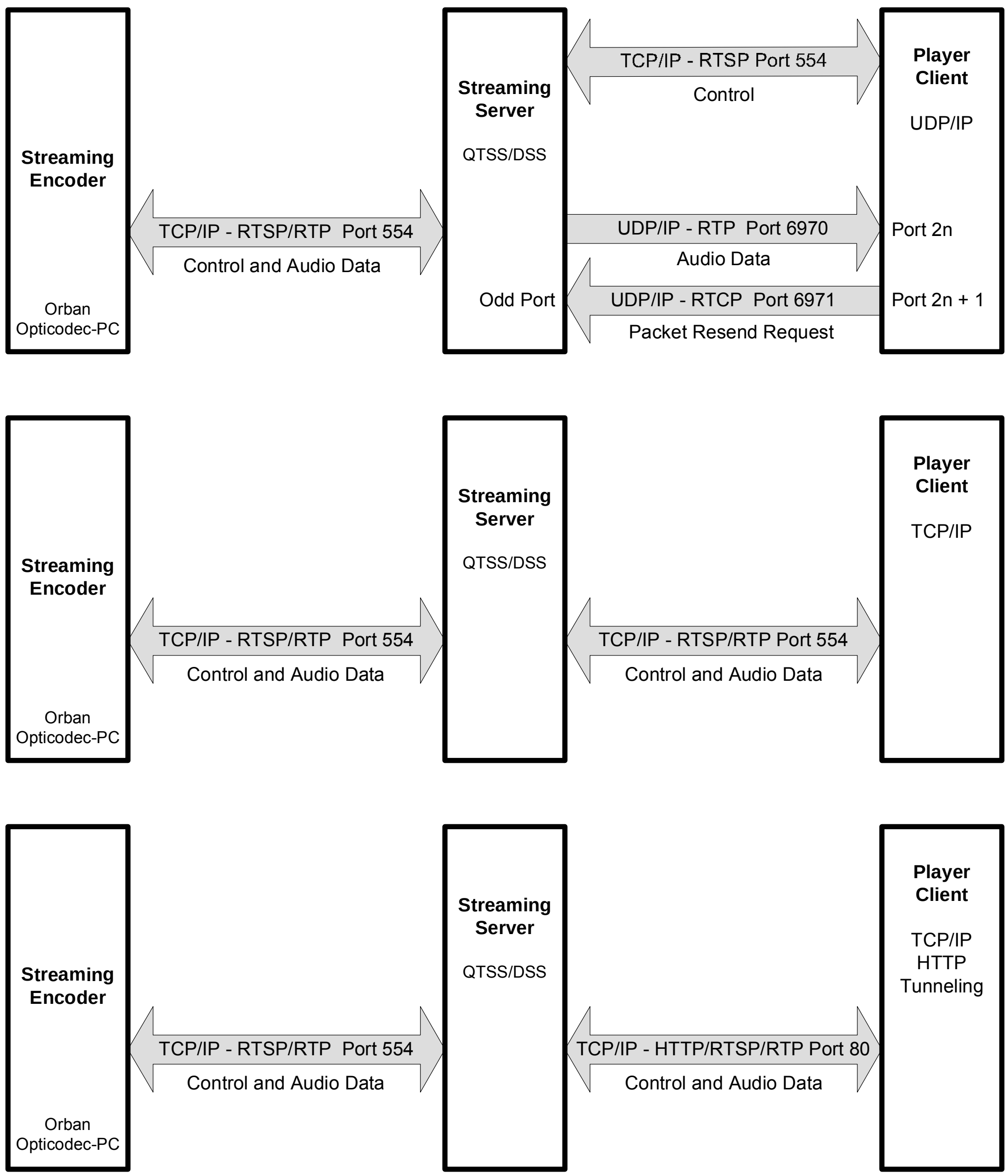


Figure 1-5: QTSS/DSS Server/Client Transports

- A special HTTP tunneling protocol that wraps RTSP/RTP packets inside HTTP packets, only supported by Apple QuickTime Player, can be used to help circumnavigate firewalls and/or routers by using the same port that web servers/browsers use. The advantage to this is that streaming protocols are still used for media content stream control and TCP/IP reliability is gained. The disadvantage is since it is less efficient, it adds server load, slight network overhead and increased bitrate.

HTTP/ICY — Streaming Servers

Using a streaming server that supports HTTP/ICY protocols such as SHOUTcast or Icecast, HTTP is fully capable of streaming media content to a client player. This is *not* the same as HTTP media file download from a web server. Basically, SHOUTcast and Icecast use HTTP to negotiate the connection between the server and clients, as well as to send metadata and to use a file transfer to send the media content.

HTTP/ICY has the following properties:

- It can be used for live and file streaming (unicast only).
- It does not leave a copy of the media content on the user's computer.
- It requires a streaming server and/or live streaming encoder.
- The stream will stop and re-buffer if data rate exceeds network connection speed.
- HTTP/ICY uses TCP/IP protocol to ensure that all streaming packets are delivered, retransmitting if necessary. TCP is optimized for guaranteed delivery of data, regardless of its format or size. For example, if your client media player realizes that it is missing a data packet from the server, it will request the server to resend that packet. Resend requests take time, take up more bandwidth, and can increase the load on the server. If the network is congested, you could begin to use more bandwidth for resends than for the media content itself. TCP is not designed for efficient real time delivery or careful bandwidth control, but for accurate and reliable delivery of every bit. Therefore, if your network bandwidth is greater than the data rate of the stream, which in most cases it is, HTTP/ICY is a very accurate way to deliver streams.
- Most firewalls, routers and network configuration schemes will pass HTTP/ICY.

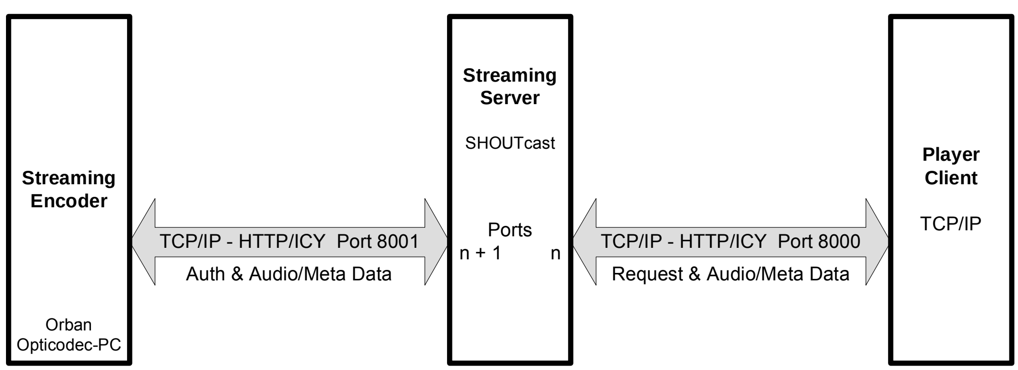


Figure 1-6: SHOUTcast DNAS Server/Client Transports

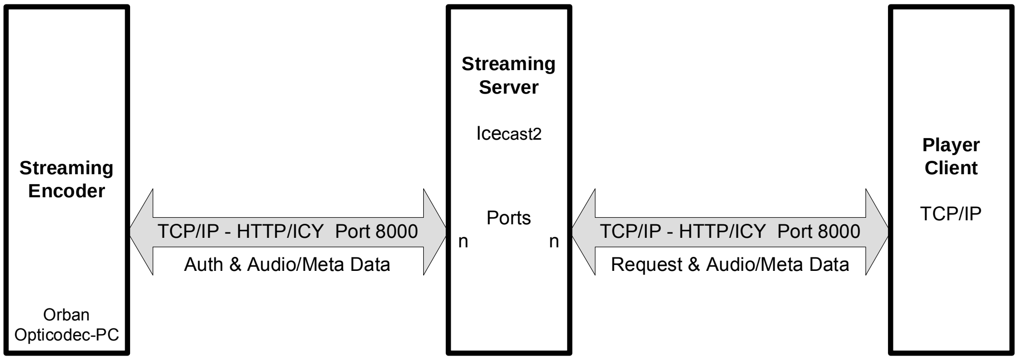


Figure 1-7: Icecast2 Server/Client Transports

HTTP — Web Servers

HTTP on web servers has the following properties:

- It is commonly called a *fast start* or *progressive download stream*.
- HTTP uses TCP/IP protocol to ensure that all streaming packets are delivered, retransmitting if necessary. TCP is optimized for guaranteed delivery of data, regardless of its format or size. For example, if your browser or client media player realizes that it is missing a data packet from the server, it will request a resend of that packet. Resend requests take time, take up more bandwidth, and can increase the load on the server. If the network is congested, you could begin to use more bandwidth for resends than for the media content itself. TCP is not designed for efficient real time delivery or careful bandwidth control, but for accurate and reliable delivery of every bit.
- HTTP does not attempt to stream in real time. To stream in real time, the bandwidth of the network must be greater than the data rate of the streaming content. If there is not enough bandwidth to transmit the streaming content in real time, streaming by HTTP allows the client to store the data locally and play the content after enough has arrived.
- HTTP is a good solution for slow, unreliable network connections; it ensures complete media content delivery.
- Most firewalls, routers, and network configuration schemes will pass HTTP.
- HTTP cannot be used for live streaming.
- HHTP provides no control over streaming content. Users cannot move to any point in a file or clip without downloading the entire file first.
- Because it uses TCP/IP, HTTP does not make efficient use of server resources. Accordingly, it does not perform as well as UDP/IP under heavy server loads.
- Beware of the copyright protection issues with on-demand file delivery, as users will be downloading your content to their computer and could easily redistribute it without your permission.

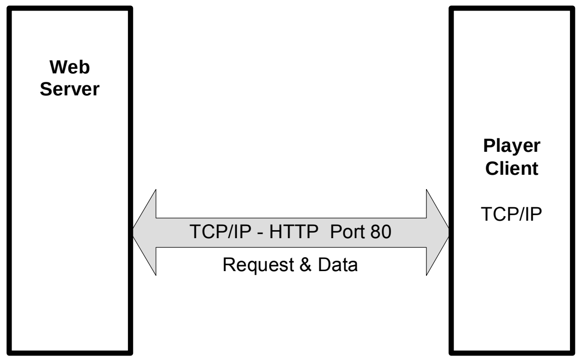


Figure 1-8: Web Server/Client Transports

Unicast

In a unicast, each user or player client initiates its own stream, resulting in several one-to-one connections between client and server, which is less efficient use of bandwidth. Many clients connected via unicast to a stream in a local network can result in heavy network traffic. However, this technique is the most reliable and most common for delivery over the Internet because no special transport support is required. For file or on-demand streams, each user can randomly access the media content, playing only the parts they want. Unicast uses either TCP/IP or UDP/IP.

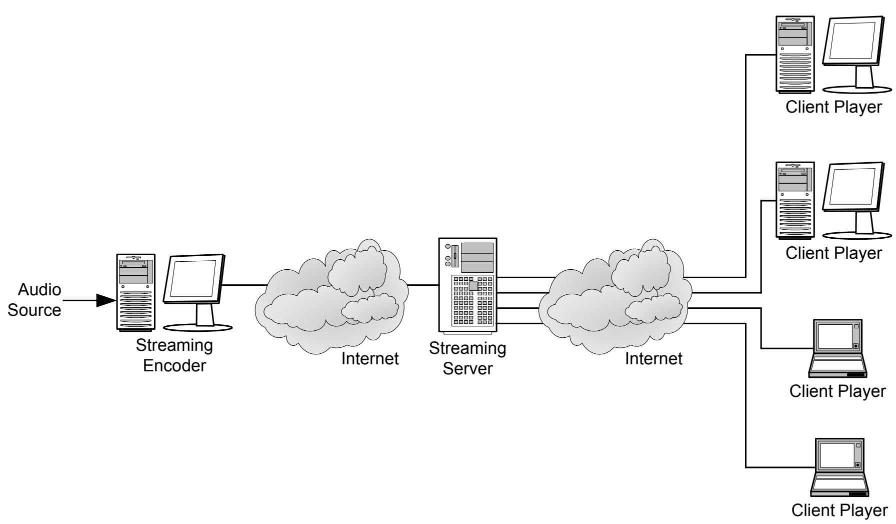


Figure 1-9: Unicast

Multicast

In a multicast, a single stream is shared among the player clients. Although this technique reduces network congestion, it requires a network that either has access to the multicast backbone (otherwise called the *Mbone*) for content generally distributed over the Internet, or is multicast-enabled for content distributed within a contained private network. Multicast streams are sent directly to a group address, such an IP multicast address, which many client computers can simultaneously access. The users of a multicast have no control over the media content. Multicasts are an efficient way to deliver the same material to a group of people over a LAN, as only one copy of the stream is sent over the network.

Multicast uses RTP and UDP/IP. Multicast is not possible using HTTP or HTTP/ICY, since it uses TCP/IP.

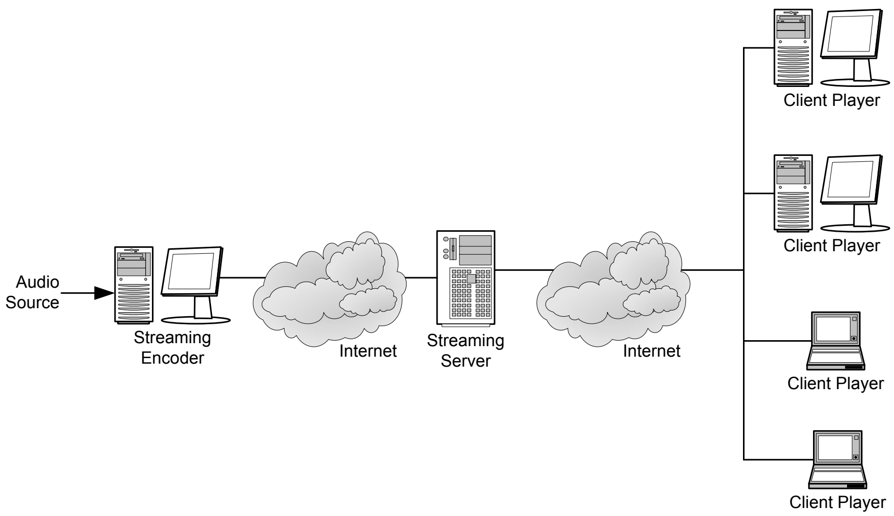


Figure 1-10: Multicast

Relay Servers

A relay server is a specially configured streaming server that listens to an incoming stream and then forwards the stream to one or more destination servers. A relay can reduce network bandwidth consumption by load balancing the stream network traffic or by separating unicast and multicast streams.

If the streaming server used supports it, Unicast and multicast can be used together with relays to create larger network infrastructures. For example, a unicast stream can be sent to a multicast server on a multicast enabled network. Alternatively, a multicast stream can be sent to a local multicast enabled LAN, and the same stream can be sent to the Internet as a unicast stream.

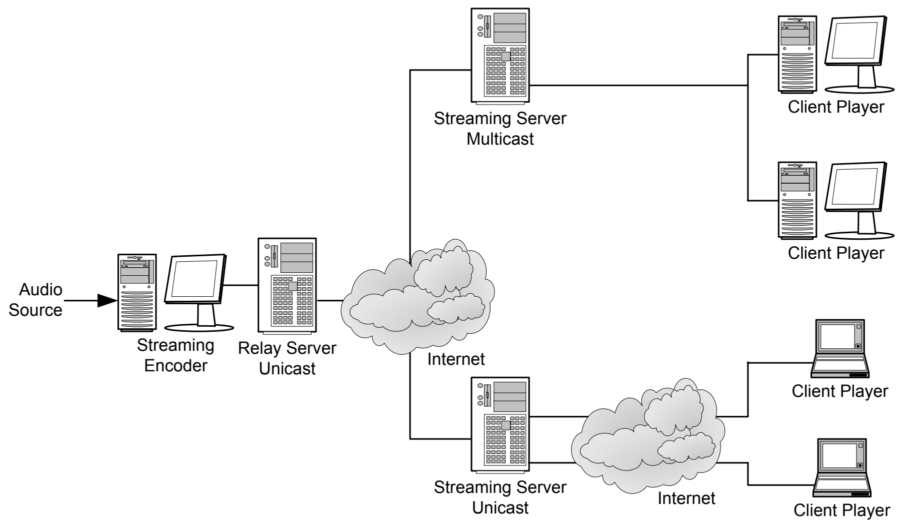


Figure 1-11: Relay Server

FUNCTION	RTSP/RTP Streaming Server	HTTP/ICY Streaming Server	HTTP Web Server
Live Streaming	■	■	
File Streaming – On-Demand	■	□	■
Playlist Streaming	■	■	
File Downloading – Progressive Download			■
File Protection	■	■	
Unicast	■	■	■
Multicast	■		
Relay	■	■	
Multiple Streams	■	□	
Authentication	■	□	□
3GPP/3GPP2 Streaming	■		
Metadata	□	■	□
ISMA Compliant	■		

Table 1-4: Summary of Server Capabilities and Compatibilities

Playing a Stream

You can play streamed media content by:

- Using a server-compatible streaming media player client and directly entering the URL of the stream.
- Using a web browser and a server-compatible streaming media player client embedded in a web page.
- Using a server-compatible hardware-based streaming media player.
- Using a server-compatible 3GPP/3GPP2 cellular streaming media player.

Section 2

Installation — Streaming Encoder

Installing Opticodec-PC

Opticodec-PC PE

For the PE version of Opticodec-PC, one Optimod-PC 1100 PCI Audio Card is required per audio program source. In a given computer, you can use as many cards as there are available PCI slots (including slots in an external PCI expansion chassis).

Multiple encoder instances (including encoders other than Opticodec-PC) can operate simultaneously to generate multiple bitrates and/or stream formats from the same program source. You may create unlimited numbers Opticodec-PC PE instances, constrained only by CPU resources. Multiple encoder instances require multi-client driver operation. Windows XP supports multi-client audio driver operation directly, while Windows 2000 requires an audio bridge application such as Ntonyx Virtual Audio Cable (http://www.ntonyx.com).

Each Opticodec-PC PE application is serialized to match the Optimod-PC card with which it will be used. It cannot be used with another Optimod-PC card. For each Optimod-PC card in a given computer, one Opticodec-PC PE application must be installed. An Opticodec-PC PE application may be moved to any computer provided its mating Optimod-PC is moved with it to the same computer.

Opticodec-PC LE

Opticodec-PC LE will operate with any quality Microsoft Windows qualified sound device. If physical audio inputs are not required (for example, by sourcing audio with a player on the same computer as the encoder) it is possible to use a virtual audio driver program, such as Ntonyx Virtual Audio Cable (http://www.ntonyx.com) to feed Opticodec-PC LE. In this way, sound device hardware, whether installed in a PCI slot or on main board, is available for other uses, such as monitoring a stream.

Only one encoder instance of Opticodec-PC LE will operate on a single computer.

Software Installation

Installation consists of:

- Installing Optimod-PC (if you are using Opticodec-PC PE, which requires the presence of Optimod-PC to start up)
- Unpacking Opticodec-PC

- Installing the Opticodec-PC encoder software on to the computer
- Optionally connecting Optimod-PC's inputs and outputs

When you have finished installing Opticodec-PC, proceed to *Configuring Opticodec-PC* on page 2-3.

1. Install Optimod-PC in your computer.

[Skip this step if you are installing an Opticodec-PC LE and are using a sound card other than Optimod-PC.]

For instructions on installing Optimod-PC, refer to the separate Operating Manual for Optimod-PC.

If you are using a sound card other than Optimod-PC, be sure that this sound card is installed and has been verified to operate correctly in Windows.

2. Unpack and inspect.

A) If you note obvious physical damage, contact the carrier immediately to make a damage claim. Included in the package are:

1 Operating Manual
1 Software CD-ROM.

B) Save all packing materials! If you should ever have to ship Opticodec-PC, it is best to ship it in the original carton with its packing materials because both the carton and packing material have been carefully designed to protect the manual and CD from damage.

C) Complete the Registration Card and return it to Orban. (please)

The Registration Card enables us to inform you of new applications, performance improvements, software updates, and service aids that may be developed, and it helps us respond promptly to claims under warranty without our having to request a copy of your original invoice or other proof of purchase. Please fill in the Registration Card and send it to us today. (The Registration Card is located after the cover page).

We do not sell our customer's names to anyone.

3. Run the installer.

A) Insert the Opticodec-PC installation CD into your computer's CD drive. The setup program will usually start up automatically.

Opticodec-PC will run only on Microsoft Windows 2000 (SP3 or higher), XP, and Server 2003. It will *not* run on earlier Windows versions, including 95, 98, 98SE, ME, and NT.

B) If the setup program does not start up:

a) Navigate to START\RUN on your computer.

b) In the Run dialog box, type `x:setup`, where "x" is the drive letter of your CD-ROM drive.

c) Click "OK."

This will install the Opticodec-PC Encoder application on your computer.

C) Answer the questions when prompted by the Orban installer.

The installer will allow you to create a desktop icon pointing to the Opticodec-PC Encoder application.

Installation on any drive is possible. We recommend installing to the default directory structure to maintain the proper directory hierarchy, *especially for multiple encoder installation*. Failure to follow this recommendation may produce unpredictable behavior.

D) After installation has completed, Opticodec-PC is ready for configuration and connection.

Software Authentication

- Opticodec-PC PE does not require software authentication. Opticodec-PC is tied to an associated Optimod-PC by serial number and can be used in or moved to any computer containing its associated Optimod-PC card.
- Opticodec-PC LE requires software authentication to activate it and to bind it to the computer hardware in which it is installed. Once the software is properly installed and runs the first time, it will guide you through the details of authentication.

  To summarize the authentication procedure:

  a) Copy the Opticodec-PC hardware ID number presented to you by the software.

  b) Submit this number by email to Orban for authentication.

  c) Orban will supply you a hardware validation number that you will enter into Opticodec-LE to complete the authentication. Once authenticated, Opticodec-PC LE will only run on the computer system to which it has been authenticated.

Configuring Opticodec-PC

This section provides the necessary information to configure , and connect the Opticodec-PC Encoder as quickly as possible. It assumes that:

- The software will be used with a TCP/IP network.
- One or more of the supported Streaming Servers, if used, are available and configured for connection from Opticodec-PC.
- Unicast or multicast operation has been decided and understood,
- Firewalls, if any, are properly configured.

For details on various streaming encoder configurations, see *Section 3*: *Configuration — Streaming Encoder* starting on page 3-1. For details on how to setup TCP/IP networking and streaming servers, see the *Section 4: Streaming Servers* starting page 4-1.

Configuration — Graphical User Interface (GUI)

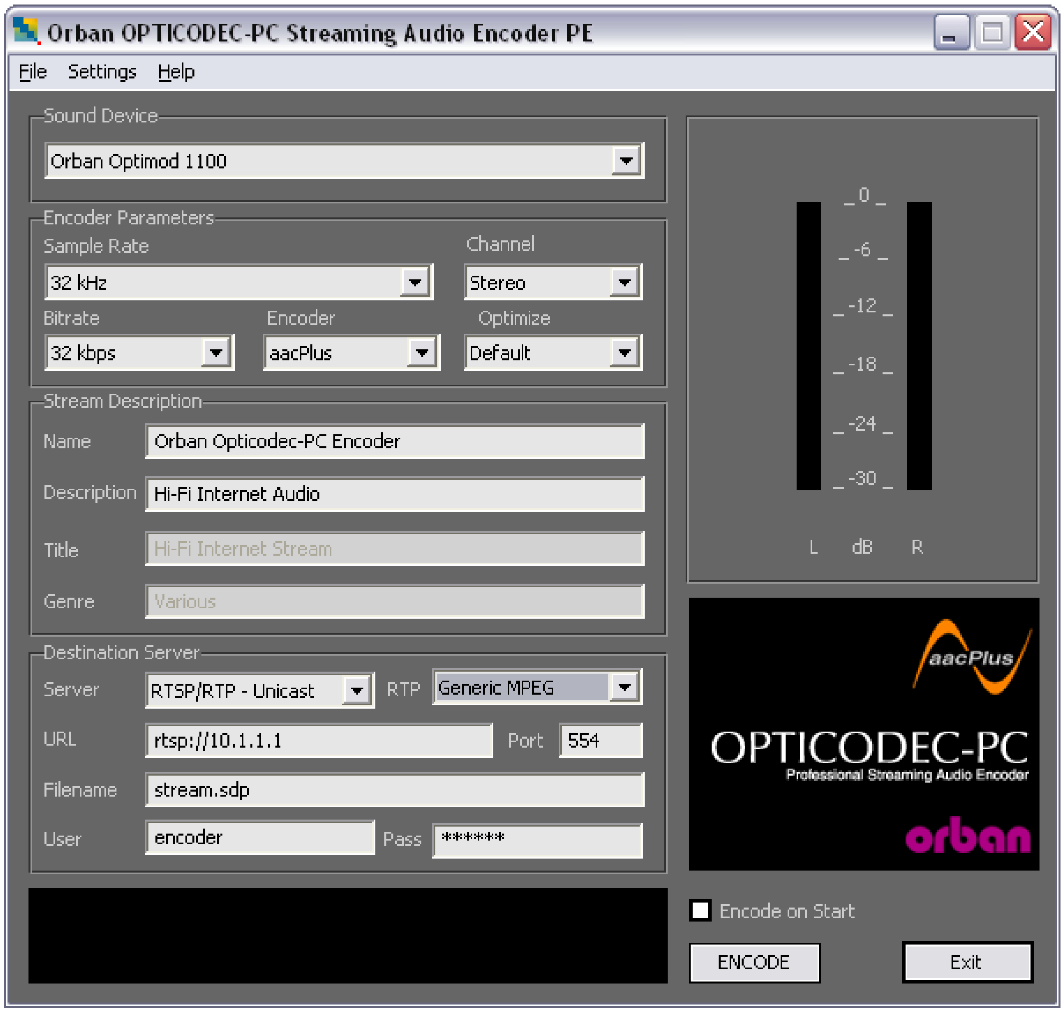


Figure 2-1: The Opticodec-PC GUI

Sound Device

The SOUND DEVICE is only available on Opticodec-PC LE. SOUND DEVICE selects the Windows sound device that will provide audio input to Optocidec-PC.

On Opticodec-PC PE, you cannot change the sound device from OPTIMOD-PC 1100

Encoder Parameters

1. Bitrate

Sets the data rate. This determines the audio quality that will be served to the intended audience. See *Table 2-1*.

Connection Speed	Bitrate Safe	Bitrate Maximum
Dial-Up 28.8kbps Modem	16kbps	20kbps
Dial-Up 56kbps Modem	32kbps	40kbps
Single ISDN 64kbps	48kbps	56kbps
Dual ISDN 128kbps	80kbps	96kbps
DSL 256kbps	128kbps	160kbps
DSL 512kbps	256kbps	384kbps
DSL 1.5Mbps	256kbps	384kbps
Cable Modem	256kbps	384kbps
Corporate LAN	128kbps	384kbps

Table 2-1: Recommended Maximum Target Bitrates

2. **Sample-Rate**

   Sets the audio sample rate. This either should be set to match the source audio or audio files or should be set lower.

   We do not recommend upsampling or setting the sample rate higher than the input rate. This does not improve quality because the original source determines the quality.

3. **Encoder**

   AAC / HE AAC / AACPLUS

   - AAC mode is recommended for bitrates of 128 kbps and higher.
   - HE AAC/aacPlus is recommended for bitrates below 128 kbps

4. **Channel**

   MONO / STEREO / STEREO V2

   - MONO/STEREO is supported by all AAC/HE AAC/aacPlus client players.
   - STEREO V2 is currently only supported by Nullsoft Winamp.

   STEREO V2 is recommended for bitrates below 48 kbps.

5. **Optimize**

   DEFAULT / VOICE

   - Use DEFAULT for mixed speech and music programming.
   - Use VOICE for programming that is exclusively speech.

Stream Description

1. **Name**

   Set the Stream Name that you want compatible player clients to display.

2. **Description**

   Set the Stream Description that you want compatible player clients to display.

3. **Title**

   Set the Metadata Title Field that you want compatible player clients to display.

4. **Genre**

   Set the Metadata Genre that you want compatible directory list servers to display.

Destination Server

1. **Server**

   Choose from the following server protocols:

   - **RTSP/RTP Unicast**
   - **RTSP/RTP Multicast**

     Although Opticodec-PC fully supports the multicast protocol, there are currently no multicast player clients available to support aacPlus/HE AAC. Apple QuickTime supports AAC only, however, aacPlus/HE AAC will play at half audio bandwidth. This will probably change in the near future.

   - **HTTP/ICY SHOUTcast**
   - **HTTP/ICY Icecast2**

2. **RTP**

   Choose from the following RTP Packet formats:

   - **MPEG4-GENERIC** is normal.
   - **MPEG4-GENERIC / MP4A-LATM** (Opticodec-PC PE only)
   - **MP4A-LATM** is used to stream to 3GPP/3GPP2 devices.

3. **URL**

   Choose the IP address or domain name of streaming server. This address or name must contain the characters `rtsp` or `http`, depending upon type of server.

4. **Port**

   Set the Port number of server used.

   The Port Number must be configured at the server.

5. **Filename**

   [For a SHOUTcast server, this field is not used.]

   - For RTSP/RTP servers, provide the name of the .sdp file.
   - For HTTP/ICY Icecast2 servers, provide the name of the mountpoint.

   The filename or mountpoint:

   - *must not* contain spaces
   - *must* end with a .sdp extension for RTSP servers
     or an .aac extension for Icecast2 servers.
   - may contain directory paths.

6. **User**

   If you are using a RTSP/RTP streaming server, provide the server encoder username.

7. **Pass**

   Provide the server encoder password for streaming server.

8. **Encode**

   Click Encode.

   This will:

   - Start the Encoder.
   - Authenticate the Encoder with the streaming server.
   - If using RTSP/RTP, send the .sdp file to the streaming server.
   - Initiate streaming.

9. **Stop**

   Click Stop.

   This will:

   - Stop the Encoder.
   - If using RTSP/RTP, the .sdp file on streaming server will be removed from the server after expiration time specified by the streaming server configuration. A reconnect is only possible after .sdp expiration.
   - Terminate streaming.

Audio Levels

Ordinarily, analog levels are calibrated to 0 VU. A properly designed audio system provides at least 20dB of headroom above 0 VU to compensate for the fact that VU meters do not indicate peaks, instead indicating average levels.

In the digital domain, it is perfectly acceptable for audio levels to reach 0 dBfs (meaning 0 dB with reference to “full-scale”: the largest peak level that the system can represent). Note that there is no headroom above 0dBfs, so any attempt to go above 0 dBfs is likely to cause audible distortion. Digital audio metering (like that found in Optimod-PC and Opticodec-PC) typically indicates true peaks, displays the exact dynamic range usage accurately, and makes it easy to prevent distortion caused by peak overloads.

> Not all sound device software can accurately meter audio peak levels. Use Opticodec-PC and/or Opticodec-PC meters to ensure that Opticodec-PC is not over-driven.

Precision peak level control is an important feature of Optimod-PC — it automatically prevents Opticodec-PC from ever being overdriven. This prevents the distortion often associated with excessive levels and keeps your audio sounding professional.

Configuration — Command Line Interface (CLI)

You can run Opticodec-PC in command-line mode. This is especially useful for automated batch controlling or automatic startup of streams. You can run several instances of Opticodec-PC PE at a time, given enough available CPU power.

Querying Audio Devices

To show the audio devices or ports available on your computer, you must run an included utility program, `showAudioInputPorts.exe`, from the command line. The program will list available devices or ports, each with a corresponding device number or audio-input-port. Use this number for the Opticodec-PC command-line syntax.

> If the Windows Preferred Audio Device is changed in the Control Panel or additional sound devices are added or removed, it is possible that the device or audio-input-port number will change. You must then rerun `showAudioInputPorts.exe` to obtain the new device or port numbers.

Opticodec-PC LE supports any Windows sound device, not just Optimod-PC. That sound device’s capabilities determine what audio sources can be used with LE. Some audio devices do not have WAV Out capability and therefore cannot route audio files between a player application and Opticodec-PC. Only physical audio inputs can be used on such devices. If WAV Out capability is required and the sound device hardware does not support it, a virtual audio driver program, such as Ntonyx Virtual Audio Cable (http://www.ntonyx.com) can feed Opticodec-PC LE. This driver will appear as another available audio device or port and will not use any hardware sound devices, leaving them free for other uses such as monitoring.

Command-Line Syntax – Unicast

```
opticodec-pc_pe <input-port-number> <sampling-frequency> <isStereo> <kbps>
<use-aacPlus> <optimizeForSpeech> <generate-LATM> <session-name> <session-info>
```

```
<session-author> <session-copyright> <rtsp-server-port> <stream-multicast>
<rtsp-server-name-or-address> <remote-file-name> <username> <password>
```

Example (Unicast to External RTSP Server):

```
opticodec-pc_pe 1 32000 1 48 1 0 0 “Stream Name” Info “” “Copyright 2004” 554 0
123.45.67.8 stream.sdp username password
```

Example (Unicast to Internal RTSP Server) (Point-to-Point):

```
opticodec-pc_pe 1 32000 1 48 1 0 0 “Stream Name” Info “” “Copyright 2004” 554 0
```

Command-Line Syntax – Multicast

```
opticodec-pc_pe <input-port-number> <sampling-frequency> <isStereo> <kbps>
<use-aacPlus> <optimizeForSpeech> <generate-LATM> <session-name> <session-info>
<session-author> <session-copyright> <rtsp-server-port> <stream-multicast>
```

Example (Multicast to Internal RTSP Server):

```
opticodec-pc_pe 1 32000 1 48 1 0 0 “Stream Name” Info “” “Copyright 2004” 554 1
```

Arguments

```
<input-port-number>
audio device number derived from running:
showAudioInputPorts.exe
1, 2, 3, 4, etc.

<sampling-frequency>
24000, 32000, 44100, 48000

<isStereo>
0 – mono, 1 – stereo

<kbps>
stream bitrate:
8, 10, 12, 16, 20, 24, 32, 40, 48, 56, 64, 80, 96, 128, 160, 192, 224, 256, 320

<use-aacPlus>
0 – AAC, 1 – HE AAC/aacPlus

<optimizeForSpeech>
0 – Normal, 1 – Voice

<generate-LATM>
RTP packet format
0 – MPEG-4-generic, 1 – MP4A-LATM

<session-name>
Stream Name of choice, enclosed in quotes if spaces are included

<session-info>
Stream Information of choice, enclosed in quotes if spaces are included

<session-author>
Stream Author of choice, enclosed in quotes if spaces are included

<stream-copyright>
Stream Copyright of choice, enclosed in quotes if spaces are included

<rtsp-server-port>
554 default, must be different for multiple multicast instances
```

```
<stream-multicast>
0 - Unicast, 1 - Multicast

<rtsp-server-name-or-address>
123.45.67.8 - ip address or domain name of streaming server

<remote-file-name>
xxxx.sdp - must end with .sdp extension and contain no spaces

<username>
Streaming server encoder username
use “” if username is not used

<password>
Streaming server encoder password
use “” if password is not used
```

Creating Batch Files

A text editor program can create batch files that use the exact same command-line syntax as that shown above. Batch files can be executed by double-clicking on them or by creating a Shortcut or Alias and place it in the Programs Startup folder.

Audio Levels

Ordinarily, analog levels are calibrated to 0 VU. A properly designed audio system provides at least 20dB of headroom above 0 VU to compensate for the fact that VU meters do not indicate peaks, instead indicating average levels.

In the digital domain, it is perfectly acceptable for audio levels to reach 0 dBfs (meaning 0 dB with reference to “full-scale”: the largest peak level that the system can represent). Note that there is no headroom above 0dBfs, so any attempt to go above 0 dBfs is likely to cause audible distortion. Digital audio metering (like that found in Optimod-PC and Opticodec-PC) typically indicates true peaks, displays the exact dynamic range usage accurately, and makes it easy to prevent distortion caused by peak overloads.

> Not all sound device software can accurately meter audio peak levels. If yours cannot, you must use a calibrated meter application like Pinguin Audio Meter.

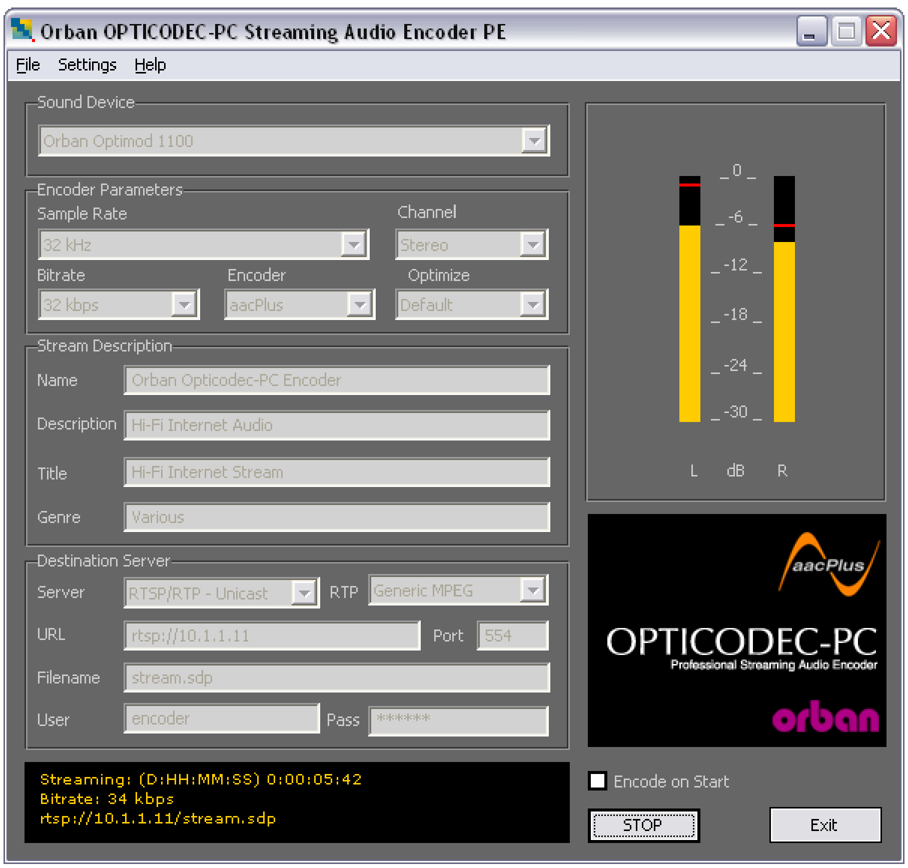
Precision peak level control is an important feature of Optimod-PC — it automatically prevents Opticodec-PC from ever being overdriven. This prevents the distortion often associated with excessive levels and keeps your audio sounding professional.

Section 3

Configuration — Streaming Encoder

Unicast — RTSP/RTP Darwin Streaming Server

Encoder/Server — Same Computer



- Any encoder running on the same computer as DSS can connect without authentication.
- A Username and Password are not required for the encoder.

Encoder Destination Server Parameters:

Server: RTSP/RTP - Unicast

URL: rtsp://127.0.0.1 or rtsp://localhost

Port: 554 (Default) – Determined by server configuration.

Username: [Leave this empty]

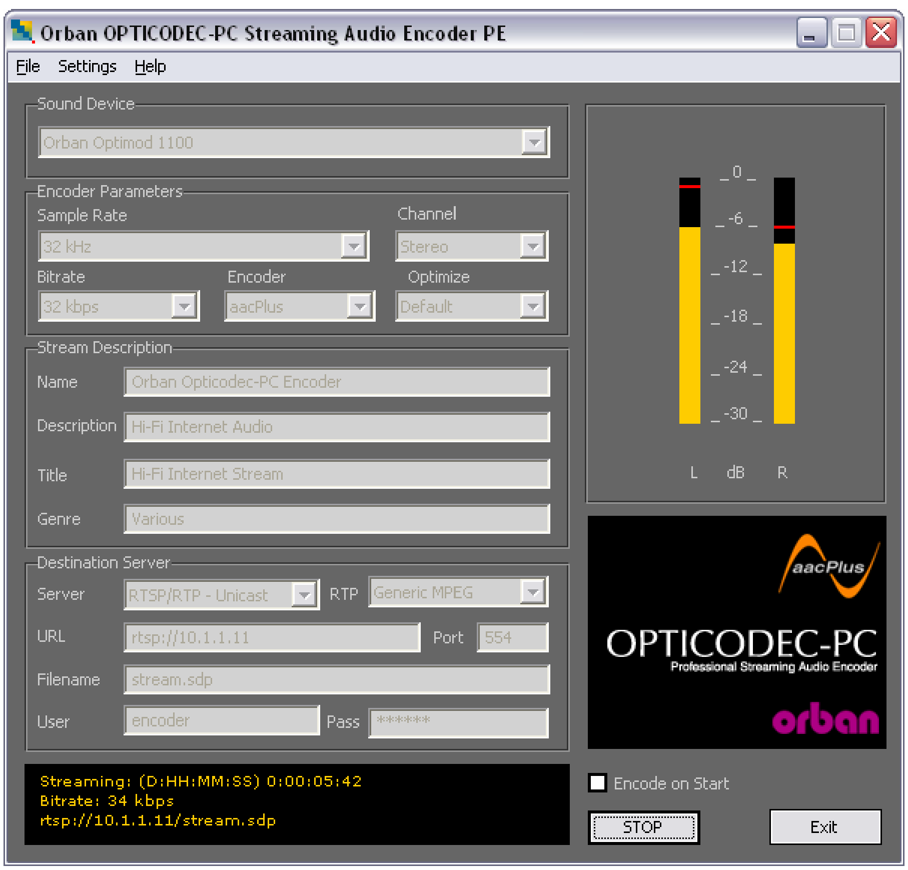
Password: [Leave this empty]

Filename: StreamName.sdp – Stream name of your choice.

The filename:

- *must not* contain spaces
- *must* end with a .sdp extension
- may contain a directory path.
- is case sensitive.

Encoder/Server — Different Computer



Any encoder running on a separate computer from QTSS/DSS requires QTSS/DSS authentication to connect.

Encoder users and access must be configured on the streaming server. Do this by adding QTSS/DSS users and allowing them server write access. If the streaming server has not been configured, refer to the *Section 4: Streaming Servers* (starting on page 4-1) for more information.

Encoder Destination Server Parameters:

Server: RTSP/RTP - Unicast

URL: rtsp://streaming.server.ip.address or
rtsp://streaming_server_domain_name

Port: 554 (Default) – Determined by server configuration.

Username: Encoder Username – Determined by server configuration.

Password: Encoder Password – Determined by server configuration.

Filename: StreamName.sdp – Stream name of your choice.

The filename:

- *must not* contain spaces.
- *must* end with a .sdp extension.
- may contain directory paths.
- Is case sensitive.

Server Connection

When Opticodec-PC Streaming Encoder negotiates a connection with the streaming server, the .sdp file is automatically uploaded to the server. If the encoder is disconnected for any reason, there is a timeout period set by the server configuration that must be expired before another connection is allowed with the same .sdp file name.

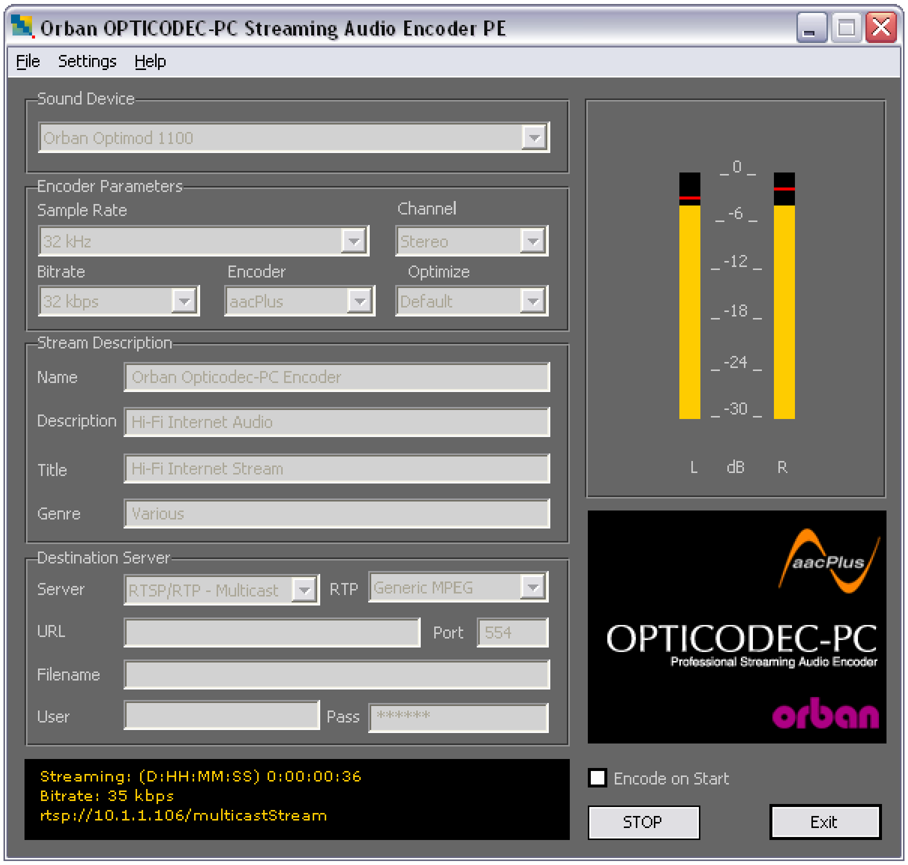
Firewall Considerations

Most hardware firewalls and routers will automatically open the necessary outbound ports. Opticodec-PC Streaming Encoder uses TCP/IP for all server connections, so there should be nothing to configure in a hardware firewall and/or router. In the event that outbound ports are blocked, or a software firewall is used, the appropriate ports will require opening.

FUNCTION	PORT	PROTOCOL	DIRECTION
Encoder – Opticodec-PC – RTSP - Default	554	TCP	Out

Table 3-1: Firewall or router configuration for RTSP/RTP Darwin Streaming Server

Multicast – RTSP/RTP Darwin Streaming Server



A server is not required for multicast operation. Opticodec-PC provides an internal multicast server. Currently, the only multicast compatible client player with Opticodec-PC is Apple QuickTime. At this time, it only supports AAC, not HE AAC/aacPlus.

Encoder Destination Server Parameters:

Server: RTSP/RTP - Multicast

URL: [Leave this empty]

Port: 554 (Default) – Determined by server configuration.

Username: [Leave this empty]

Password: [Leave this empty]

Filename: [Leave this empty]

The URL that users must enter into their multicast compatible client players will be indicated in the status region of Opticodec-PC Encoder. The URL will be in the form: `rtsp://ip.address/multicastStream`.

Note the case sensitivity in the URL.

Multicast Relay

Relaying a multicast stream requires a multicast enabled network and an RTSP/RTP server such as QuickTime Streaming Server or Darwin Streaming Server. Configure Opticodec-PC to supply a unicast stream to the streaming server; the streaming server handles the multicast.

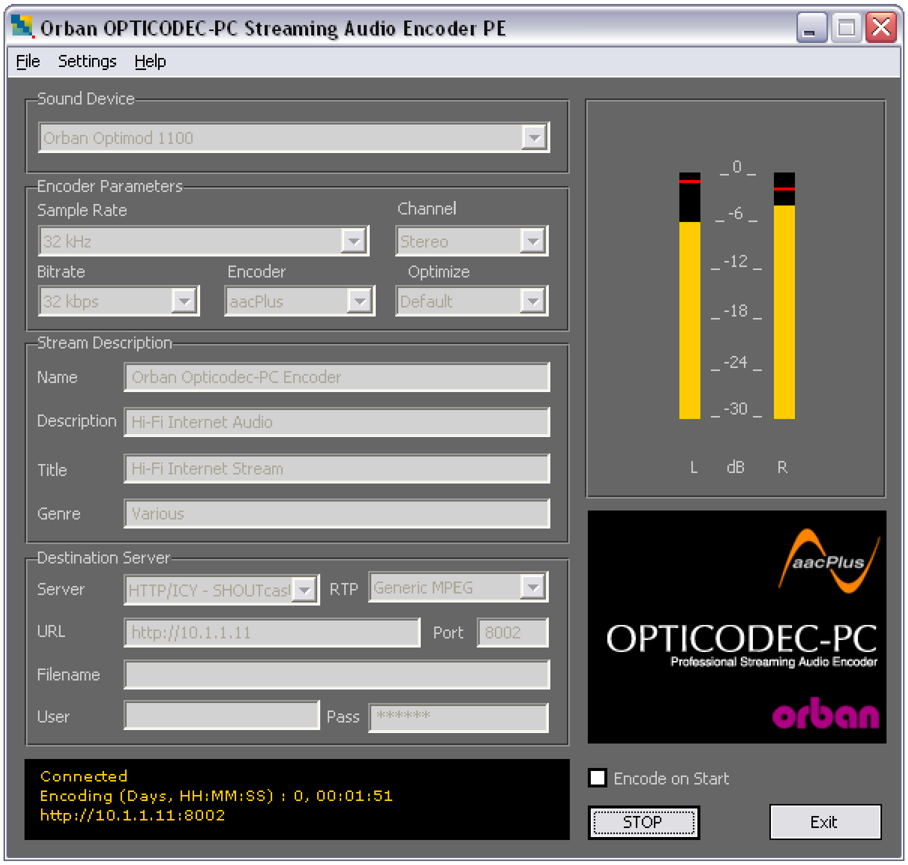
Multicast Addresses

Multicast uses Class D addresses, 224.0.0.0 through 239.255.255.255. If you are to multicast on the Internet, you must allocate these addresses through IANA. Opticodec-PC will dynamically assign a multicast address to multicast sessions. To view this address, you can use a TCP/UDP port program such as TCPView.

Firewall Considerations

Whenever using multicast, through a firewall or router, it must be configured to pass the multicast address range. Many firewalls and routers are configured by default to block the entire multicast address range. This is an important consideration if streaming on Intranets or the Internet.

Unicast — HTTP/ICY SHOUTcast



Because Nullsoft Winamp, the primary player for Icecast2 streams, supports for HE AAC/aacPlus v2, we recommend using a Channel setting of Stereo v2. This will provide the best codec performance at low bitrates.

Encoder Destination Server Parameters:

Server: HTTP/ICY SHOUTcast

URL: http://streaming.server.ip.address or http://streaming_server_domain_name

Port: 8000 (default) – Determined by server configuration.

Username: [Leave this empty]

Password: Encoder (Source) Password – Determined by server configuration.

Filename: [Leave this empty]

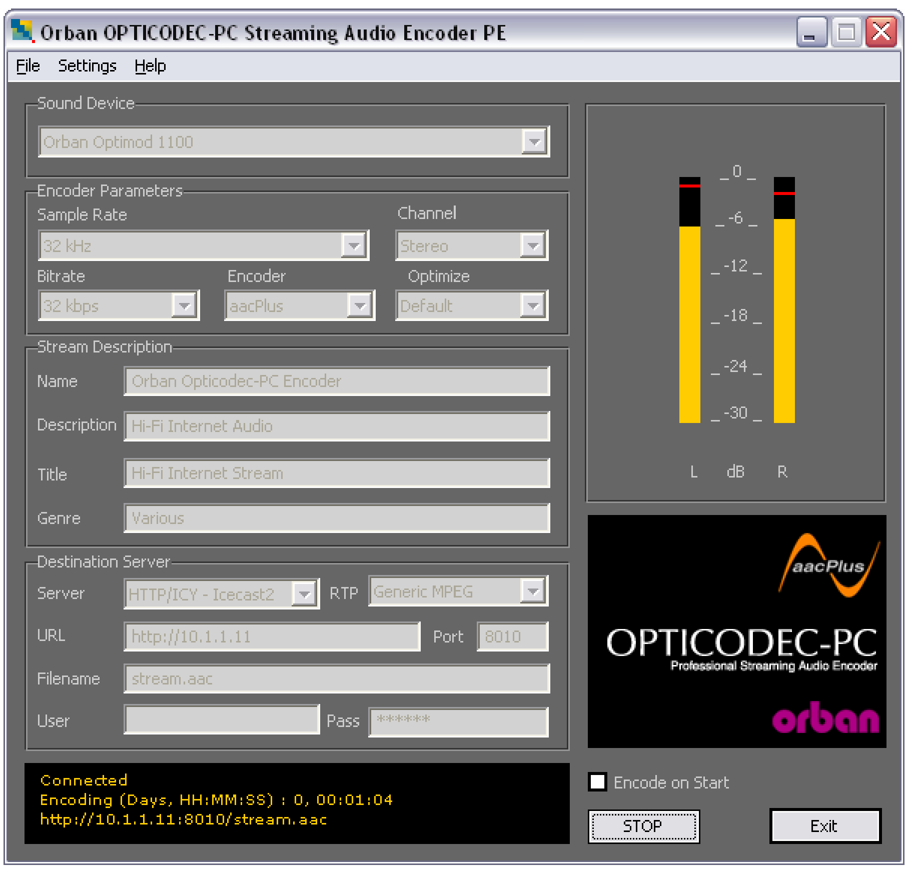
Firewall Considerations

Most hardware firewalls and routers will automatically open the necessary outbound ports. Opticodec-PC Streaming Encoder uses TCP/IP for all server connections, so there should be nothing to configure in the firewall and/or router. In the event that outbound ports are blocked or a software firewall is used, the appropriate ports will require opening.

FUNCTION	PORT	PROTOCOL	DIRECTION
Encoder – Opticodec-PC – RTSP – Default	8000	TCP	Out

Table 3-2: Firewall or router configuration for HTTP/ICY SHOUTcast

Unicast — HTTP/ICY Icecast2



Because Nullsoft Winamp, the primary player for Icecast2 streams, supports for HE AAC/aacPlus v2, we recommend using a Channel setting of Stereo v2. This will provide the best codec performance at low bitrates.

Encoder Destination Server Parameters:

Server: HTTP/ICY Icecast2

URL: http://streaming.server.ip.address or http://streaming_server_domain_name

Port: 8000 (default) – Determined by server configuration.

Username: [Leave this empty]

Password: Encoder (Source) Password – Determined by server configuration.

Filename: mountpoint.aac – Mountpoint name of your choice.

The filename:

- *must not* contain spaces
- *must* end with a .aac extension
- may contain a directory path.
- is case sensitive.

Firewall Considerations

Most hardware firewalls and routers will automatically open the necessary out-bound ports. Opticodec-PC Streaming Encoder uses TCP/IP for all server connections, so there should be nothing to configure in the firewall and/or router. In the event that outbound ports are blocked or a software firewall is used, the appropriate ports will require opening.

FUNCTION	PORT	PROTOCOL	DIRECTION
Encoder – Opticodec-PC – RTSP – Default	8000	TCP	Out

Table 3-3: Firewall or router configuration for HTTP/ICY Icecast2

Section 4

Streaming Servers

Introduction

Orban Opticodec-PC Streaming Encoder supports several different streaming server platforms. Choosing which platform to use will depend upon several things, including intended audience, supported client players, supported client player features, compatibilities, server features, and administration complexity. You could even consider using more than one server platform.

Darwin Streaming Server (DSS)

Darwin Streaming Server is a full featured, scaleable, enterprise-class streaming media server that can stream both live streams encoded with Orban Opticodec-PC Streaming Encoder and files produced by the Orban Opticodec-PC File Encoder. It uses standards-based RTSP/RTP and HTTP protocols. It is open-source, based on the same code base as the Apple QuickTime Streaming Server, provides a high level of customizability by allowing code manipulation, and runs on a variety of computer platforms. Both DSS and QTSS are built on a core server that provides state of the art quality of service features and support for the latest digital media standards, MPEG-4 and 3GPP.

Supported Protocols

- **RTSP over TCP**. The Real Time Streaming Protocol (RTSP) is a client-server multimedia presentation control protocol that provides efficient delivery of streamed multimedia over IP networks. RTSP provides a basis for negotiating unicast and multicast transport protocols, such as RTP, and negotiates codecs in a way that is independent of file format. It works well for large audiences as well as single-viewer media-on-demand. RFC 2326 defines the IETF standard for RTSP. Both QuickTime Player 6 and RealPlayer 10 support this protocol.
- **RTP over UDP**. The Realtime Transport Protocol (RTP) is a packet format for multimedia data streams. RTP is used by many standard protocols, such as RTSP (for streaming applications) and SDP (for multicast applications). RTP provides the data delivery format for RTSP and SDP. RFC 1889 defines the IETF proposed standard for RTP. Both QuickTime Player 6 and RealPlayer 10 support this protocol.
- **RTP over Reliable UDP**. If an RTP player client requests it, the server sends RTP packets using Reliable UDP. Reliable UDP is a set of quality-of-service en-

hancements (such as congestion control tuning improvements, retransmit, and thinning server algorithms) that improve the ability to present a good quality RTP stream to RTP clients even in the presence of packet loss and network congestion.

- **RTSP/RTP in HTTP (tunneled)**. Firewalls often prevent users on private IP networks from receiving RTSP/RTP streams. On private networks, an HTTP proxy server is often configured to provide users with indirect access to the Internet. To reach such clients, QuickTime Player 6 supports the placement of RTSP and RTP data in HTTP requests and replies, allowing viewers behind firewalls to access RTSP/RTP streams through HTTP proxy servers. Both QuickTime Player 6 and RealPlayer 10 support this protocol.
- **RTP over RTSP (RTP over TCP)**. Certain firewall designs and other issues may require a server to use alternative means to send data to clients. RFC 2326 allows RTSP packets destined for the same control endpoint to be packed into a single lower-layer protocol data unit (PDU), encapsulated into a TCP stream, or interleaved with RTP and RTCP packets. Interleaving complicates client and server operation and imposes additional overhead, so it should only be used if RTSP is carried over TCP. When the transport is RTP, RTCP messages are also interleaved by the server over the TCP connection. By default, RTCP packets are sent on the first available channel numbered higher than the RTP channel. The client may request RTCP packets on another channel explicitly. This is done by specifying two channels in the interleaved parameter of the transport header. RTCP is used for synchronization when two or more streams are interleaved. In addition, this provides a convenient way to tunnel RTP/RTCP packets through the TCP control connection when required by the network configuration, and to transfer them onto UDP when possible. Both QuickTime Player 6 and RealPlayer 10 support this protocol.
- **HTTP/ICY (SHOUTcast/Icecast)**. This implementation of HTTP/ICY is Icecast1. This protocol has been deprecated and Icecast2 supersedes it. Orban Opticodec-PC Streaming Encoder does not currently support it.

DSS Installation — Windows

This information will guide you through a typical basic Darwin Streaming Server (DSS) software installation for unicast operation on a Microsoft Windows operating system. This is the most common streaming server configuration.

> Installation, setup, and configuration of server software is not recommended for computer novices because of complexity and potential security risks. Proceed with caution. Knowledge of basic Command Prompt operation is assumed.

For other server configurations, operating systems, and more details, consult the *Darwin Streaming Server Administration Manual*, available at:

http://developer.apple.com/darwin/projects/streaming/qtss_admin_guide.pdf

Depending upon the application and/or client player capacity of the streaming system you intend to serve, Darwin Streaming Server requirements will vary.

Because Darwin Streaming Server is available for Microsoft Windows, it is possible to run the Opticodec-PC encoder on the same computer system. This is only recommended for small LAN or WAN applications. For larger Internet streaming applications, it is always preferable to use separate encoder and server computers to optimize reliability, redundancy, and system architecture.

Operating System Requirements

Microsoft Windows 2000 Professional/Server, XP Professional, or 2003 Server

Software Requirements

ActivePerl 5.8 or above:
http://www.activeperl.com/Products/ActivePerl/

Darwin Streaming Server 5.0 or above:
http://developer.apple.com/darwin/projects/streaming/

Optional Useful Network Utilities

NetPerSec: http://www.pcmag.com/article2/0,1759,1681,00.asp

NetLimiter: http://www.netlimiter.com/index.php

TCPView: http://www.sysinternals.com/ntw2k/source/tcpview.shtml

Ethereal: http://www.ethereal.com

Installation

1. **Install ActivePerl.**

   A) Download ActivePerl 5.8 or above for Microsoft Windows.

   http://www.activeperl.com/Products/ActivePerl/

   B) Save the .msi Installer to any directory.

   C) Run the .msi installer by double-clicking on it.

   D) Accept all defaults.

2. **Install Darwin Streaming Server (DSS)**

   A) Download Darwin Streaming Server

   http://developer.apple.com/darwin/projects/streaming/

   B) Save the distribution self-extracting .exe file into any directory.

   C) Run the distribution self-extracting .exe file into a temporary directory.

   D) Run `install.bat` by double-clicking on it.

   > `Install.bat` runs in a command box. It copies all files into `C:\Program Files\Darwin Streaming Server` and starts the Darwin Streaming Server as a Service. By default, the Service is configured as Automatic Startup.

   The installer prompts for the Administrator Username.

   E) Supply the Administrator Username.

The installer prompts for the Administrator Password.

F) Supply the Administrator Password.

The installer adds the Administrator Username and Password.

Do not close the command box.

Minimizing the command box is OK. Closing the Command Box terminates Perl and will not allow the subsequent operations to complete installation and configuration.

G) Open a Web Browser to: `http://127.0.0.1:1220`

The DSS Login will appear.

H) Log in as Administrator using the Username and Password assigned previously.

On the first login, DSS will complete its configuration.

I) DSS Admin will prompt for the MP3 Broadcast Password. Enter it.

Although this function is not used for the RTSP/RTP version of Opticodec-PC Encoder, it should be entered and secured. This is for streaming Shoutcast/Icecast HTTP compliant streams and may be changed at any time.

J) DSS Admin will prompt for SSL. Enable it if required.

Enable this option only if you administer DSS remotely, require extra security, and have a valid SSL certificate installed for secure remote administration. Your browser must also support SSL.

K) DSS Admin will display the default Media path. Accept the default unless you wish to change to a different Media path directory.

If you change from the default Media path, it is necessary to create the directory if it does not already exist.

L) DSS Admin will prompt for Streaming on Port 80. Use this option to stream using the HTTP protocol with encapsulated RTP over Port 80.

Do not use streaming on Port 80 on the same computer or IP address as a web server, as Port 80 will conflict with HTTP web server traffic. Currently, Streaming on Port 80 only works with QuickTime Player and allows streaming through firewalls that block Port 554 or UDP packets.

DSS is now ready to accept connections from Players, such as QuickTime, and RealPlayer.

3. Test the Darwin Streaming Server.

Using QuickTime, try to play one of the test files that installed with the server software installation.

This can be done on the local or a remote computer.

A) Start QuickTime.

B) From the File menu, Open the URL using the following syntax:

`rtsp://dss.server.ip.address/sample_100kbit.mp4`

where `dss.server.ip.address` is the IP address or hostname of DSS.

The test file should play the QuickTime logo with audio and video.

The supplied test files will not work with RealPlayer, as they are in an unsupported format.

Security: Access and Authentication

A certain level of security is inherent in real-time streaming because content is delivered only as the client player needs it and no media files remain afterwards. However, you should address other security issues. These include access control for encoders and client players, if needed.

By default, DSS has no authentication configured for encoder-server connections. There are two encoder-server configurations that determine access and/or authentication to allow you to create the session description protocol file (.sdp file) required for live streams. The .sdp file provides information about the format, timing, and authorship. DSS access and authentication are configured using text files, called *access files*, in very much the same way that an Apache web server is configured.

DSS access and authentication are not configured using the DSS Remote Administration browser interface.

DSS can also be configured to control client player access to live streams as well as streamed media files. Currently, only QuickTime Player 6 above supports this; RealPlayer 10 does not.

For access control to work, an access file must be present in the DSS media directory. If an access file is not present, all player clients are allowed to access the media and live streams in the directory, and all encoders on other computers will have no write access and thus will not connect.

Different encoder-server configurations require slightly different setups:

Opticodec-PC Encoder and DSS on the Same Computer

Any encoder running on the same computer as DSS can connect without authentication. DSS does not require or use authentication in this configuration. A Username and Password are not required for the encoder to have write access to create the .sdp file in the media directory, and therefore no further configuration is necessary.

Opticodec-PC Encoder Destination Server Parameters:

URL: rtsp://127.0.0.1 or rtsp://localhost
Port: 554
Username: [Leave this empty]
Password: [Leave this empty]
Filename: StreamName.sdp

The filename:

- may include a directory path
- must contain no spaces
- must end with an .sdp extension.

Opticodec-PC Encoder and DSS on Different Computers

Any encoder running on a different computer from DSS requires DSS authentication to connect. You have two choices: (1) allowing unlimited access from any encoder user or (2) limiting access from encoder users. Implement DSS authentication by creating an access file and editing it accordingly.

If you allow unlimited access, any user is allowed server write access in the access file. Only use unlimited access for internally controlled networks such as LAN/WAN configurations. Because there is no authentication, any Orban Opticodec-PC or Apple QuickTime Broadcaster can connect to the server and use unauthorized streaming bandwidth.

To limit access, you must add encoder users. Do this by adding DSS users and configuring the access file for write access (see *Access Control* on page 3-7). Opticodec-PC can then create the .sdp file in the directory you specified (see **Filename** in *Opticodec-PC Encoder Destination Server Parameters* on page 3-5).

Opticodec-PC Encoder Destination Server Parameters – Any User

URL: rtsp://streaming.server.ip.address *or* rtsp://streaming_server_domain_name
Port: 554
Username: [Leave this Empty]
Password: [Leave this Empty]
Filename: StreamName.sdp
The filename:

- may include a directory path
- must contain no spaces
- must end with an .sdp extension.

Opticodec-PC Encoder Destination Server Parameters – Limit User

URL: rtsp://streaming.server.ip.address *or* rtsp://streaming_server_domain_name
Port: 554
Username: username
Password: password
Filename: StreamName.sdp
The filename:

- may include a directory path
- must not contain any spaces
- must end with an .sdp extension.

Authenticated Client Player Access

To configure client player access for authenticated live streams and/or streamed media files, you must create an access file in the media directory you wish to secure. If the access file is shared with encoder access entries, you must create or add the appropriate entries to the access file. The directory that is used to store .mp4 and .mov streamed media files and/or .sdp files for live streams can contain other directories and each directory can contain its own access file. Currently, only QuickTime Player 6 (and above) supports authenticated client player access; RealPlayer 10 does not.

To disable authentication for a media directory, edit the file accordingly. If live streams are used, remove the access file or rename it.

Access Control

Allow Write Access

By default, DSS does not contain a `qtaccess` file. You must create a `qtaccess` file and place it in the DSS media directory, Windows default, `C:\Program Files\Darwin Streaming Server\Movies`. The file must be named `qtaccess` and saved as plain text without formatting.

Use a text editor program such as Notepad.

`qtaccess` file to allow any encoder access to DSS:

```
<Limit WRITE>
require any-user
</Limit>
require any-user
```

`qtaccess` file to limit encoder access to DSS:

```
<Limit WRITE>
require user username
</Limit>
require any-user
```

`username` is an encoder username of your choice.

Allow Authenticated Client Player Access

By default, DSS does not contain a `qtaccess` file. For access control to work, an access file must be present in the DSS media directory, Windows default, `C:\Program Files\Darwin Streaming Server\Movies`. If an access file is not present, all clients are allowed to access the media in the directory, and all encoders on other computers will have no write access. The file must be named `qtaccess` and saved as plain text without formatting.

Use a text editor program such as Notepad.

`qtaccess` file for authenticated client player access:

```
<Limit WRITE>
require user username
</Limit>
require valid-user
```

`username` is an encoder username of your choice.

Access File Reference

The qtaccess file is a text file in the DSS Media directory, subdirectory, or set of files in subdirectories. The qtaccess file will protect all media files in the directory with the qtaccess file and its subdirectories.

It is important to remember that the qtaccess file can control both encoder and client player access, depending upon the file's location.

The parameters in the file can contain some or all of the following:

```
AuthName <message>
AuthUserFile <user filename>
AuthGroupFile <group filename>
require user <username1> <username2>
```

```
require group <groupname1> <groupname2>
require valid-user
require any-user
<Limit Write>
</Limit>
```

Terms not in angle brackets are keywords. Anything in angle brackets is information you supply. Do not include the angle brackets.

- `AuthName <message>`
  Name of the authentication domain (optional). This is the text that users see when the login window appears. If your message includes spaces, make sure you enclose the entire text within quotation marks. This is not supported in the current login window dialog box, so this option will have no effect on client systems.

- `AuthUserFile <user filename>`
  Name and path of the user file. If it is not specified, the default qtusers file will be used. For Windows, it is `C:\Program Files\Darwin Streaming Server\qtusers`.

- `AuthGroupFile <group filename>`
  Name and path of the groups file (optional). If it is not specified, the default qtgroups file will be used. For Windows, it is `C:\Program Files\Darwin Streaming Server\qtgroups`. A group file is optional. If you have many users, instead of listing each user it may be easier to set up one or more groups and then to enter the group names.

- `AuthScheme`
  Authentication scheme, either digest or basic, providing the ability to specify the authentication scheme on a directory-by-directory basis.

- `require user <username1> <username2>`
  Limits access to the media directory to the specified list of users. (A "user" is authorized to log in and access the media file, live stream, or live encode.) The username must be in the user file you specified. You can also specify `valid-user`, which designates any valid user.

- `require group <groupname1> <groupname2>`
  Limits access to the media directory to the specified list of groups defined in the groups file. The group and its members must be listed in the groups file specified.

- `require valid-user`
  Limits access to the media directory to any valid users defined in the qtusers file. The statement `require valid-user` specifies that any authenticated user in the `qtusers` file can have access to the media files. If this tag is used, the server will prompt users for an appropriate username and password.

- `require any-user`
  Provides unlimited access to the media directory. Allows any user to access media without providing a username or password. This is the most common type of access for streaming media.

- <Limit Write>
  Provides access control for encoder write access to DSS. The `require` statements above can be placed within the <Limit Write></Limit> tags. This construct is used for the RTSP announce protocol to limit the users or encoders, that can broadcast through the server.
- </Limit>

To Add a User:

A) Open a Command Prompt.

B) Change the directory to `C:\Program Files\Darwin Streaming Server`.

C) Run: `qtpasswd -f <user filename> <username>`

Example: `qtpasswd -f qtusers encoder`

where `qtusers` is the DSS user filename and `encoder` is the username.

D) DSS will prompt for user password. Enter it.

E) To add additional Users, repeat steps (B) through (D).

Be careful when using `qtpasswd` in with the -c option. The `qtusers` file also contains the account information of the “admin” user and it is possible to overwrite the original `qtusers` file.

To Change a User Password:

A) Open a Command Prompt.

B) Change the directory to `C:\Program Files\Darwin Streaming Server`.

C) Run: `qtpasswd <username>`

Example: `qtpasswd encoder`

D) DSS will prompt for user password. Enter it. The password you enter replaces the previous password.

Group File Reference (Optional)

The `qtgroups` file is a text file in the DSS root directory, Windows default, `C:\Program Files\Darwin Streaming Server\qtgroups`. If you have many users, rather than listing each user it may be easier to set up one or more groups and then enter the group names.

```
<groupname1>:<username1> <username2>
<groupname2>:<username3> <username4>
```

To Add a Group:

A) Open the groups file with a text editor.

Use a text editor program such as Notepad.

B) Add the appropriate groupname(s) and username(s) to the file.

Example:

```
encoders:encoder1 encoder2
players:player1 player2
```

C) Save the file as plain text without formatting.

To Delete a User from a User or Group File:

A) Open the user file with a text editor.

Use a text editor program such as Notepad.

B) Delete the username and encrypted passwords line from the user file.

C) Delete the username from the group file, if used.

Remote Administration

A Perl script controls DSS Remote Administration. By default, and to supply extra security, DSS Remote Administration does not start automatically on boot. To allow DSS Remote Administration to be available after any reboot, it is recommended, but not required, that a Shortcut or Alias be created and placed in the Programs Startup folder. Without this Shortcut, DSS Remote Administration will be unavailable until `streamingadminserver.pl` is executed manually by double-clicking it.

In other words, create a Shortcut for:

```
C:\Program Files\Darwin Streaming Server\streamingadminserver.pl
```

and include it in:

```
C:\Documents and Settings\All Users\Start Menu\Programs\Startup
```

Other Configuration Options

There are several other server configuration options available by editing the `streamingserver.xml` file located in the Darwin Streaming Server root directory. This file is thoroughly commented.

Use a text editor program such as Notepad.

Firewall Considerations

DSS is a very secure, robust streaming server designed to be connected directly to the Internet if required. It is not necessary or recommended to operate DSS from behind a firewall or router using Network Address Translation (NAT). This makes things much more complicated than they need to be. However, if DSS is located behind a firewall or router and access from the Internet is required, specific ports need to be opened in the firewall to allow Real Time Streaming Protocol (RTSP) requests from users, encoded audio from the encoder and outbound streams to clients, and DSS Remote Administration on the local network and the Internet.

FUNCTION	PORT	PROTOCOL	DIRECTION
Encoder – Opticodec-PC – RTSP - Default	554	TCP	In
Encoder – Opticodec-PC – HTTP - Default	8001	TCP	In
Player-Control – RTSP	554	TCP	In
Player-Media – RTSP/RTP/UDP	6970-6999	UDP	Out / In
Player-Media – RTSP/RTP/TCP Interleave	554	TCP	Out
Player-Media – HTTP Tunneling	80	TCP	Out
Player-Media – HTTP/ICY - Default	8000	TCP	Out
DSS Remote Admin – HTTP	1220	TCP	In
DSS Remote Admin – HTTPS	1240	TCP	In

Table 4-1: Firewall configuration for various protocols and servers

If the firewall or router is using NAT, it is necessary to configure DSS with the IP address of the Internet side of the router or firewall for UDP streams to work properly. Do this by editing the `alt_transport_src_ipaddr` option in the `streamingserver.xml` file located in the Darwin Streaming Server root directory.

The default value is empty:

```
<PREF NAME="alt_transport_src_ipaddr" ></PREF>
```

Change it to contain the Internet IP address (for example):

```
<PREF NAME="alt_transport_src_ipaddr" >123.45.67.8</PREF>
```

DSS must be restarted in order for this to take effect.

If you do not have a static IP address, then DSS needs to be reconfigured every time your IP address changes.

It is wise to have a static IP address when streaming.

SHOUTcast™ DNAS (Distributed Network Audio Server)

SHOUTcast™ DNAS (Distributed Network Audio Server) is a free, downloadable streaming media server that can stream live streams encoded with Orban Opticodec-PC Streaming Encoder and files produced by the Orban Opticodec-PC File Encoder, including stream-related metadata. SHOUTcast DNAS requires one server instance and port pair per stream. SHOUTcast DNAS offers a web based stream redirector, such that if a stream URL is entered into a web browser, the stream can be played by the user's clicking on the appropriate link, provided that the user has an AAC/HE AAC/aacPlus player client such as Winamp installed and configured correctly.

SHOUTcast DNAS is available for Microsoft Windows and various UNIX platforms and uses the popular HTTP/ICY protocol. The Windows version offers both a command line (CLI) and graphical user interface (GUI). SHOUTcast is a product of Nullsoft, Inc., which also provides the popular free Windows Winamp player client.

Supported Protocol

- **SHOUTcast HTTP/ICY over TCP**. The SHOUTcast HTTP/ICY protocol is based on the same TCP HTTP protocol used for web servers. The encoder and player client negotiation, authentication, metadata, and media data are specific to SHOUTcast. This protocol has the advantage of working through most firewalls easily.

Installation – Microsoft Windows

This information will guide you through a typical SHOUTcast DNAS GUI software installation on a Microsoft Windows operating system and will provide you with any Orban Opticodec-PC specific information.

Installation, setup, and configuration of server software is not recommended for computer novices because of complexity and potential security risks. Proceed here with caution. Knowledge of basic Command Prompt operation is assumed.

For other server configurations, operating systems, and more details, consult the documentation available at:

http://www.shoutcast.com/support/docs/

Complete SHOUTcast DNAS documentation is provided in the README.TXT file that is installed with DNAS in the same directory.

Depending upon the application and/or client player capacity of the streaming system you intend to serve, SHOUTcast DNAS requirements will vary.

Because SHOUTcast DNAS is available for Microsoft Windows, it is possible to run the Opticodec-PC encoder on the same computer system. This is only recommended for small LAN or WAN applications. For larger Internet streaming applications, it is always preferable to use separate encoder and server computers to optimize reliability, redundancy, and system architecture.

Operating System Requirements

Microsoft Windows 2000 Professional/Server, XP Professional, or 2003 Server

Software Requirements

SHOUTcast DNAS 1.9.4 or above:
http://www.shoutcast.com/download/serve.phtml

Optional Useful Network Utilities

NetPerSec: http://www.pcmag.com/article2/0,1759,1681,00.asp

NetLimiter: http://www.netlimiter.com/index.php

TCPView: http://www.sysinternals.com/ntw2k/source/tcpview.shtml

Ethereal: http://www.ethereal.com

Installation

1. **Download SHOUTcast DNAS.**

   http://www.shoutcast.com/download/license.phtml

   A) Accept the license agreement.

   B) Save the SHOUTcast WIN32 Console/GUI server file to any directory.

2. **Install SHOUTcast™ DNAS.**

   A) Run the distribution self-extracting .exe to install.

   B) Accept all the defaults.

   Do not check both GUI and console versions to install; otherwise, only the console version will install.

   The installer will automatically install the DNAS, an uninstaller, and Start Menu Shortcuts.

   DNAS installation is complete.

Configuration — Single Stream

1. **Start SHOUTcast DNAS.**

   A) Go to the Start Menu / SHOUTcast DNAS

   B) Click the EDIT CONFIG in the menu bar.

   A text editor will appear with the Configuration file for the server.

2. **Edit the SHOUTcast DNAS Configuration File.**

   A) Edit the items under the REQUIRED STUFF header.

   The default values are:

```
MaxUser=32
Password=changeme
PortBase=8000
```

   The PortBase value and value + 1 must be available. e.g.: 8000/8001

   It is unnecessary for these to be even numbers. However, even numbers are generally used.

   Except for port 80, we recommended using ports above 1024.

   If port 80 is used, it should get through even the tightest firewalls. If a web browser can connect to a web site, then a streaming player client should connect.

   B) Edit any other relevant items.

   The sc_serv.ini file is thoroughly commented and describes the various server options.

   C) Save the file when done.

   D) Restart the server. The new changes will take effect.

   DNAS installation is now complete.

3. **Create a Shortcut for Auto Startup (optional).**

   A) Create a Shortcut for:

```
C:\Program Files\SHOUTcast\sc_serv.exe
```

   B) And include it in:

```
C:\Documents and Settings\All Users\Start Menu\Programs\Startup
```

Configuration — Multiple Streams

Each Opticodec-PC stream requires a separate instance of SHOUTcast DNAS. Each instance must be configured for a different base port. This can be managed by using .bat files that call SHOUTcast DNAS sever and separate configuration files. The .bat files can instruct multiple servers to start multiple instances at the same time. They can also auto start upon Windows startup.

1. **Edit the SHOUTcast DNAS Configuration File(s).**

   A) Open the configuration file with a text editor.

   Use a text editor program such as Notepad.

   B) Edit the items under the REQUIRED STUFF header.

   The default values are:

```
MaxUser=32
Password=changeme
PortBase=8000
```

The PortBase value and value + 1 must be available for each server instance. e.g.:
8000/8001, 8002/8003, 8004/8005, 8006/8007…
It is unnecessary for the pairs to increase in increments of 2.
It is unnecessary for the first number in the pair to be even. However, even numbers are generally used.

Except for port 80, we recommend using ports above 1024.

If port 80 is used, it should get through even the tightest firewalls. If a web browser can connect to a web site, then a streaming player client should connect.

C) Edit any other relevant items.

The sc_serv.ini file is thoroughly commented and describes the various server options.

D) When finished, save the file with a unique filename having a file extension of .ini for each server instance, e.g.:

```
sc_serv1.ini, sc_serv2.ini…
```

This will be the filename specified in the .bat file created later.

2. Create .bat Files.

A) Open a text editor program such as Notepad.

B) Create an entry or entries to start SHOUTcast DNAS by calling the appropriate .ini file.

Example .bat file to start one SHOUTcast DNAS Server:

```
sc_serv.exe sc_serv1.ini
```

Example .bat file to start four SHOUTcast DNAS Servers:

```
sc_serv.exe sc_serv1.ini
sc_serv.exe sc_serv2.ini
sc_serv.exe sc_serv3.ini
sc_serv.exe sc_serv4.ini
```

C) When finished, save the file with a .bat file extension in the same directory as SHOUTcast DNAS.

D) Double-clicking the .bat file will start the SHOUTcast DNAS servers.

E) DNAS installation is now complete.

3. Create Shortcuts for Auto Startup (optional).

A) Create a Shortcut for:

C:\Program Files\SHOUTcast*name-of-.bat-file(s)*

B) And include it in:

```
C:\Documents and Settings\All Users\Start Menu\Programs\Startup
```

Server User Interface

This is the SHOUTcast DNAS graphical user interface.

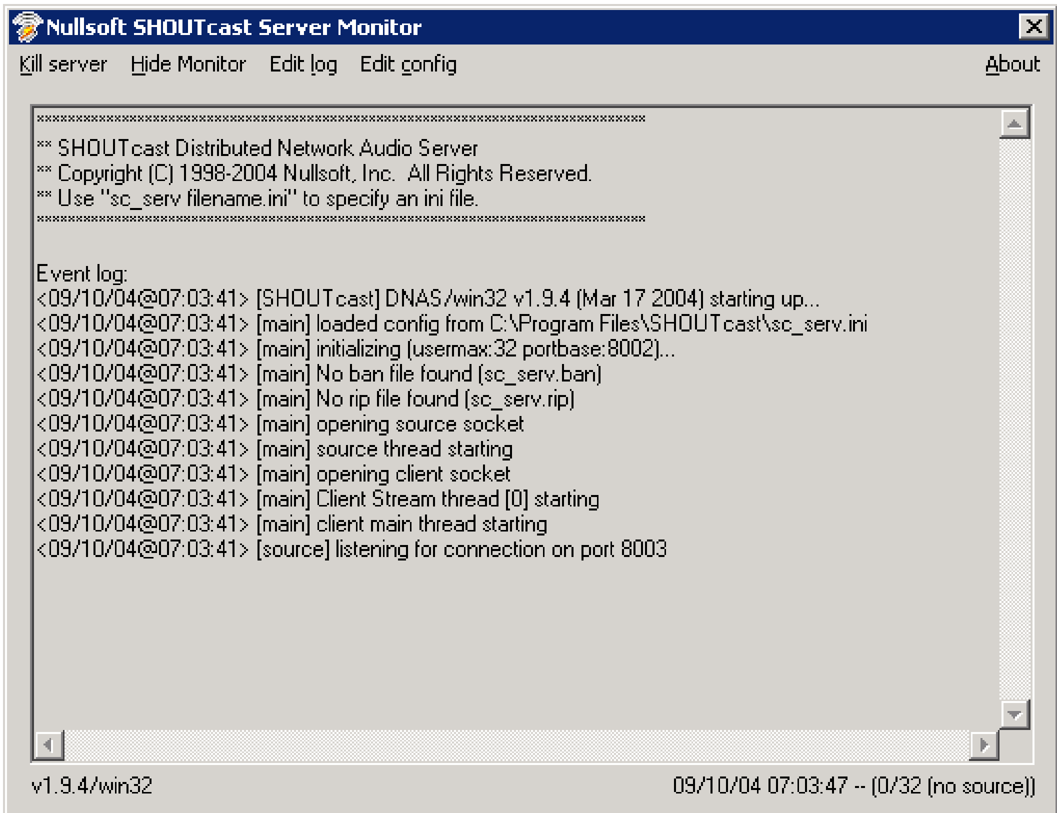


Figure 4-1: SHOUTcast DNAS graphical user interface

Remote Administration

SHOUTcast DNAS provides remote administration via a web browser.

A) Open a web browser to:

```
http://serverIP-or-domain:port
```

B) Log in using the admin password that is specified in the config file.

Firewall Considerations

SHOUTcast DNAS is a very secure, robust streaming server that is designed to be connected directly to the Internet if required. Operating SHOUTcast DNAS from behind a firewall or router using NAT is not required or recommended. However, if SHOUTcast DNAS is located behind a firewall or router, and access from the Internet is required, specific ports need to be opened and forwarded in the firewall to allow inbound encoder connections and outbound HTTP requests from player clients to stream media data to them.

FUNCTION	PORT	PROTOCOL	DIRECTION
Encoder – Opticodec-PC – HTTP - Default	8001	TCP	In
Player-Media – HTTP/ICY - Default	8000	TCP	Out

Note: Ports shown in this chart are default values. Actual ports used must be configured in the firewall and/or router.

Table 4-2: Firewall considerations for SHOUTcast DNAS

Port forwarding must also route the incoming HTTP requests from player clients to the IP address of the SHOUTcast DNAS.

For multiple streams, the necessary port pairs need to be opened and forwarded accordingly.

Icecast2

Icecast2 is a free downloadable open-source streaming audio server capable of streaming both live streams encoded with Orban Opticodec-PC Streaming Encoder and files produced by Orban Optidodec-PC File Encoder, including stream-related metadata. Icecast2 also offers additional server security by using encrypted passwords for encoder access. It is available for Microsoft Windows and various UNIX platforms and uses the popular HTTP/ICY protocol. The Windows version offers both a command line (CLI) and graphical user interface (GUI). Icecast2 supports mountpoints, thus allowing multiple streams on a single server using the same port. Icecast2 produces streams compatible with the popular free Windows Winamp player client.

Supported Protocol

- **Icecast2 HTTP/ICY over TCP**. The Icecast HTTP/ICY protocol is based on the same TCP HTTP protocol used for web servers. The encoder and player client negotiation, authentication, and metadata and media data is specific to Icecast2. This protocol has the advantage of working through most firewalls easily.

Installation — Microsoft Windows

This information will quickly guide you through a typical Icecast2 GUI software installation on a Microsoft Windows operating system and provide you with any Orban Opticodec-PC specific information.

Installation, setup, and configuration of server software is not recommended for computer novices because of complexity and potential security risks. Proceed with caution. Knowledge of basic Command Prompt operation is assumed.

For other server configurations, operating systems, and more details, consult the documentation available at:

http://www.icecast.org/docs.php

Complete Icecast2 documentation is also provided in the icecast2.chm file, which is installed with Icecast2 in the same directory and is also accessible under the About/Help menu.

Depending upon the application and/or client player capacity of the streaming system you intend to serve, Icecast2 requirements will vary.

Since Icecast2 is available for Microsoft Windows, it is possible to run the Opticodec-PC encoder on the same computer system. This is only recommended for small LAN or WAN applications. For larger Internet streaming applications, it is always preferable to use separate encoder and server computers to optimize reliability, redundancy, and system architecture.

Operating System Requirements

Microsoft Windows 2000 Professional/Server, XP Professional, or 2003 Server

Software Requirements

Icecast2 2.0.1 or above:
http://www.icecast.org/download.php

Optional Useful Network Utilities

NetPerSec: http://www.pcmag.com/article2/0,1759,1681,00.asp

NetLimiter: http://www.netlimiter.com/index.php

TCPView: http://www.sysinternals.com/ntw2k/source/tcpview.shtml

Ethereal: http://www.ethereal.com

Installation

1. **Download Icecast2.**

   http://www.icecast.org/download.php

   A) Download the Windows version.

B) Save the Icecast2 server file to any directory.

2. Install Icecast2.

A) Run the distribution self-extracting .exe to install.

B) Accept all the defaults.

The installer will automatically install Icecast2, an uninstaller, and a Start Menu Shortcut.

Icecast2 installation is complete.

Configuration

1. Start Icecast2.

A) Go to the Start Menu / Icecast 2

B) Click Configuration / Edit Configuration in the menu bar.

A text editor will appear with the icecast.xml Configuration file for the server.

2. Edit the Icecast Configuration File.

A) Edit the items under the `<authentication>` header.

The default values are:

```
<source-password>hackme</source-password>
<admin-password>hackme</admin-password>
```

B) If a port other than the default is to be used, edit the item under the `<listen-socket>` header.

The default value is:

```
<port>8000</port>
```

Make sure the correct uncommented listen socket is edited. Otherwise, there will be no change to the listen socket.

C) Edit any other relevant items.

The icecast.xml file is commented and describes the various server options. More thorough information can be found in the About / Help menu, or in the icecast2.chm file in the Icecast2 Win32 directory.

D) Save the file when done.

Icecast2 installation is now complete.

3. Create Shortcut for Auto Startup (optional).

A) Create a Shortcut for:

C:\Program Files\Icecast2 Win32\Icecast2.exe

B) And include it in:

C:\Documents and Settings\All Users\Start Menu\Programs\Startup

Server User Interface

This is the Icecast2 server graphical user interface. It shows the Server Status and Source Level Stats tabs.

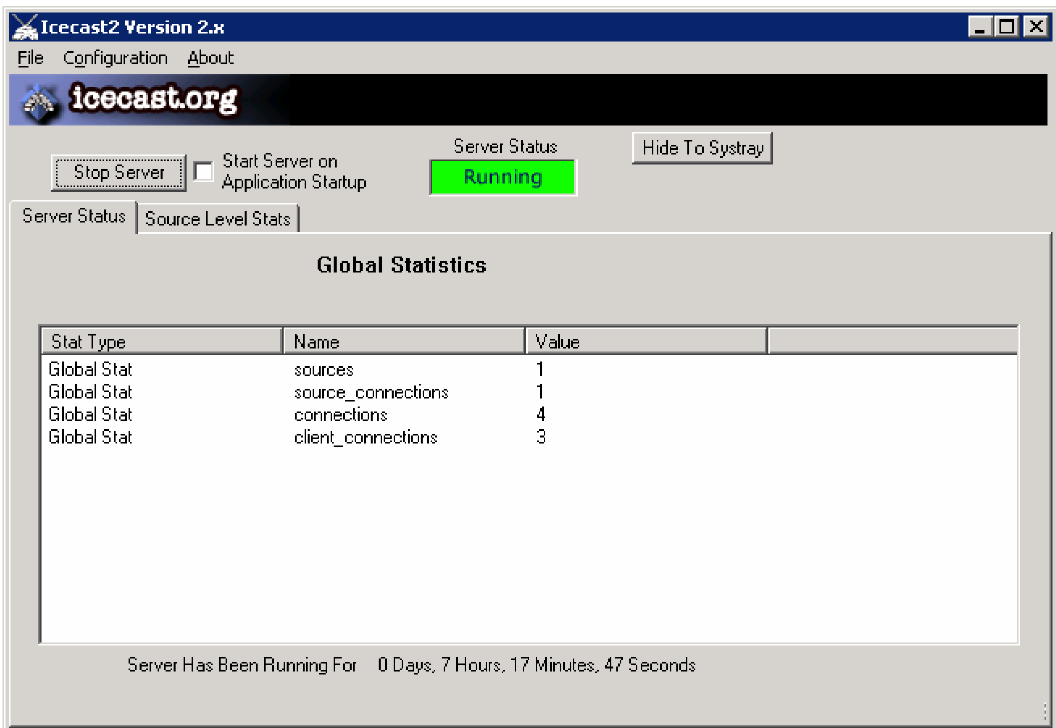
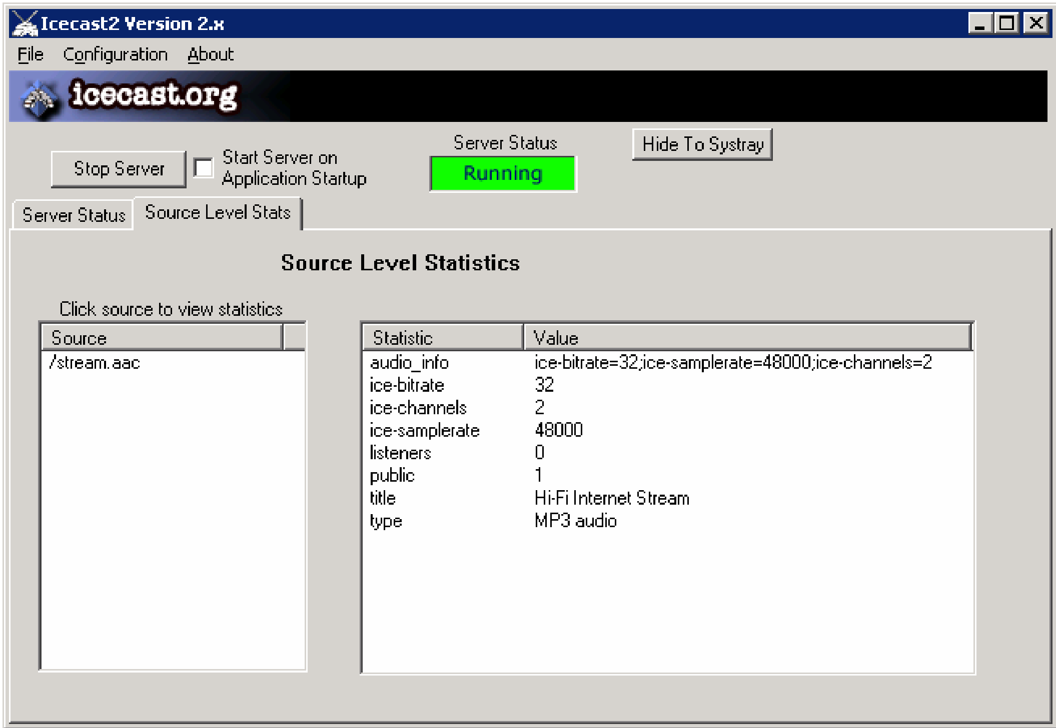


Figure 4-2: Icecast Server Status Tab



Note: AAC/HE AAC/aacPlus streams display as MP3 audio, since Icecast2 uses the same content type to stream both AAC and MP3 streams.

Figure 4-3: Icecast Source Level Stats Tab

Streaming Files

Icecast2 provides a means of streaming static on-demand AAC/HE AAC/aacPlus files produced with Opticodec-PC File Encoder. This is not the same as a web server. A web server downloads the file to the client player computer, even though the file will play as it is downloading. Conversely, Icecast2 streams the file. It does not download the file.

If the default directory is used, then files may be placed in:

```
C:\Program Files\Icecast2 Win32\web
```

They may be streamed to a client player such as Winamp by opening:

```
http://ip-address-or-domain:port/filename.aac
```

Remote Administration

Icecast2 provides remote administration via a web browser. Open a web browser to:

```
http://serverIP-or-domain:port/admin/stats.xsl
```

Log in using the admin password that is specified in the config file.

Firewall Considerations

Icecast2 is a very secure, robust streaming server that is designed to be connected directly to the Internet if required. Operating Icecast2 from behind a firewall or router using NAT is not necessary or recommended. However, if Icecast2 is located behind a firewall or router and access from the Internet is required, specific ports need to be opened and forwarded in the firewall to allow inbound encoder connections and outbound HTTP requests from player clients to pass to and from the server.

FUNCTION	PORT	PROTOCOL	DIRECTION
Encoder – Opticodec-PC – HTTP - Default	8000	TCP	In
Player-Media – HTTP/ICY - Default	8000	TCP	Out

Note: Ports shown in this chart are default values. Actual ports used must be configured in the firewall and/or router.

Table 4-3: Firewall considerations; Icecast2

Port forwarding must also route the incoming HTTP requests from player clients to the IP address of the Icecast2 server.

Multiple Servers

It is possible to operate several type of streaming servers on a single computer in order to cover all possible combinations of features and client player compatibilities. If your server is single-homed (that is, having a single IP address), make sure each server has its own port allocation so there are no conflicts. If your server is multi-homed and each streaming server is on a separate IP address, then port allocation is not an issue.

Section 5

Client Players

Comparisons

Opticodec-PC AAC/HE AAC/aacPlus streams can be played on various software and hardware client players, including 3GPP/3GPP2 wireless devices. Software client players are available for different computer operating systems or platforms. All player clients are not created equal; they support different features and connection protocols. This chart is a comparison of some of the available players and their supported protocols. More players are adding support for AAC/HE AAC/aacPlus as users discover its superior sound quality and bandwidth efficiency. Open-source players are also available, although currently they do not offer reliable performance.

Software Client Player Platforms

PLAYER	Windows	Windows CE	Macintosh	Linux
QuickTime 6	■		■	
RealPlayer 10	■		■	■
Winamp 5.05	■			
Coding Technologies Player		■		

Table 5-1: Software Client Player Platforms

Software Client Player AAC/HE AAC/aacPlus Codecs

PLAYER	AAC	HE AAC/aacPlus	PCM/WAV
QuickTime 6	■		
RealPlayer 10	■	■	■
Winamp 5.05	■	■	■
Coding Technologies Player	■	■	

Table 5-2: Software Client Player AAC/HE AAC/aacPlus Codecs

Software Client Player File Formats

PLAYER	ADTS .aac/.acp	MPEG-4 .mp4/.m4a	3GPP .3gp	3GPP2 .3g2
QuickTime 6	■	■	■	■
RealPlayer 10	■	■	■	■
Winamp 5.05	■			
Coding Technologies Player		■		

Table 5-3: Software Client Player File Formats

Software Client Player Streaming Protocols

	UNICAST				MULTICAST
PLAYER	RTSP/RTP UDP Reliable	RTSP/RTP TCP Interleave	HTTP/RTP TCP Tunneling	HTTP/ICY TCP	RTSP/RTP UDP
QuickTime 6	■		■		■
RealPlayer 10	■	■	Proprietary		Proprietary
Winamp 5.05				■	
Coding Technologies				■	

a) Note: The protocols indicated here represent support for AAC/HE AAC/aacPlus streams.

Table 5-4: Software Client Player Streaming Protocols

Different streaming protocols suit different network conditions. On a lossless network such as a well designed LAN/WAN, any of the protocols will work with little performance difference. Multicast will provide the absolute least amount of network traffic, but any firewalls or routers through which the stream passes must be multicast enabled. Unicast RTSP/RTP/UDP will provide the least overhead and should operate without routing problems, especially if all users are on the same side of any firewall or router, if used. RTSP/RTP/UDP can, however, sometimes be problematic on networks with packet loss and through some NAT firewall or routers. Under these circumstances, we advise configuring your client player to use RTSP/RTP/TCP, or HTTP/RTP/UDP if it is enabled on the streaming server.

Real Networks RealPlayer 10

RealPlayer is used to play RTSP/RTP streams served from an RTSP/RTP server such as Darwin Streaming Server. RealPlayer has HTML embedded player capability for embedding players into web pages and/or creating custom players. RealPlayer currently does not support private authenticated streams supported by Darwin Streaming Server.

Figure 5-1: RealPlayer 10 GUI

Direct URL Entry and Play

1. Start RealPlayer.

A) Go to the START MENU / REALPLAYER or double-click the RealPlayer Windows Taskbar Icon.

B) Click FILE / OPEN in the menu bar.

An Open dialog box will appear.

2. Enter the URL of the stream.

A) If the file or live stream is served from the server root media directory, the syntax is:

```
rtsp://ip.address.or.domain:rtsp_port/stream.sdp
```

Examples:

```
rtsp://123.45.67.8:554/live.sdp

rtsp://opticodec.net:554/live.sdp
```

B) If the file or live stream is served from a directory within the server root media directory, the syntax is:

```
rtsp://ip.address.or.domain:rtsp_port/directory/stream.sdp
```

Examples:

```
rtsp://123.45.67.8:554/source1/live.sdp

rtsp://opticodec.net:554/source2/live.sdp
```

3. Enter the URL of the file stream.

A) If the file stream is served from the server root media directory, the syntax is:

```
rtsp://ip.address.or.domain:rtsp_port/file.mp4
```

Examples:

```
rtsp://123.45.67.8:554/file.mp4

rtsp://opticodec.net:554/file.mp4
```

B) If the file stream is served from a directory within the server root media directory, the syntax is:

```
rtsp://ip.address.or.domain:rtsp_port/directory/file.mp4
```

Examples:

```
rtsp://123.45.67.8:554/source1/file.mp4

rtsp://opticodec.net:554/source2/file.mp4
```

Metafile/Playlist Files

RealPlayer can play live or file streams that are referenced in metafiles or playlist files. This is useful for creating aliases or shortcuts that the user simply double-clicks to automatically launch RealPlayer and start playing the stream, distribution to other users, or for use as streaming links on a web pages. Metafiles or playlist files are text files that simply contain information about the stream, such as the stream URL, and depending upon the type of metafile or playlist file, some optional information to control the player.

There are two common types of metafiles or playlist files for use with RealPlayer, .ram and .rpm. These are text files and the syntax for them is exactly the same. The .ram file is used to launch RealPlayer and the .rpm file is used to start an embedded RealPlayer from within a web page. Microsoft Windows systems register both of these file types upon a RealPlayer installation. Because both the .ram and .rpm extensions are exclusive to RealPlayer, it is therefore a sure way to always launch RealPlayer by using either of them. It is highly unlikely that another program would have appropriated the .ram and .rpm file extensions, preventing RealPlayer from being associated incorrectly. Additional features of the .ram and .rpm files include the ability to pass titling information and additional instructions to RealPlayer.

1. Create .ram File(s).

A) Open a text editor program such as Notepad.

B) Create an entry containing the streaming URL.

```
rtsp://ip_address_or_domain:rtsp_port/stream.sdp
```

Example .ram files:

```
rtsp://123.45.67.8:554/live.sdp

rtsp://opticodec.net:554/live.sdp

rtsp://opticodec.net:554/live1.sdp?rpcontextheight=240&rpcontextwidth=320
&rpconteturl="http://www.opticodec.net/playlist1.html"&title="OPTICODEC.net"
&author="Hi-Fi Audio for the Internet"&copyright="&#169;2004, Orban, Inc."
```

C) Save the file with a .ram file extension when done.

Double-clicking the .ram file will launch RealPlayer, connect, and play the stream.

2. **Create .rpm File(s).**

A) Open a text editor program such as Notepad.

B) Create an entry containing the streaming URL.

```
rtsp://ip_address_or_domain:rtsp_port/stream.sdp

rtsp://ip.address.or.domain:rtsp_port/stream.sdp?parameter=value

rtsp://ip.address.or.domain:rtsp_port/stream.sdp?parameter=value
&parameter=value&parameter=value...
```

Example .rpm files:

```
rtsp://123.45.67.8:554/live.sdp

rtsp://opticodec.net:554/live.sdp
```

C) Save the file with a .rpm file extension when done.

The .rpm file is for use with HTML embedded RealPlayers within a web page. It can only be tested on a web server. HTML embedded players are covered in *Embedded Players* on page 5-9.

Additional RealPlayer Parameters and Values

Separate the first parameters from the stream URL with a question mark (?). To set two or more parameters for the same stream or file, precede the second and all subsequent parameters with ampersands (&) instead of question marks.

Parameter	Value	Default	Function
author	*text*	(none)	Indicates the clip author.
clipinfo	title=*text*| artist name=*text*| album name=*text*| genre=*text*| copyright=*text*| year=*text*| cdnum=*number*| comments=*text*	(none)	Gives extended clip information.

Parameter	Value	Default	Function
copyright	*text*	(none)	Gives the copyright notice
end	*hh:mm:ss.x*	(none)	Ends the clip at the specified point.
mode	normal|theater|toolbar	normal	Opens RealPlayer in one of three initial playback modes.
rpcontextheight	*pixels*	media height	Sets the related info pane's height.
rpcontextparams	*parameters*	(none)	Adds parameters to rpcontexturl.
rpcontexttime	*dd:hh:mm:ss.x*	0	Specifies a time at which an HTML page displays in the related info pane.
rpcontexturl	*URL*|_keep	(none)	Displays the specified URL in the related info pane.
rpcontextwidth	*pixels*	330	Sets the related info pane width.
rpurl	*URL*	(none)	Gives a URL for the media browser.
rpurlparams	*parameters*	(none)	Appends parameters to rpurl.
rpurltarget	_rpbrowser|*name*	_rpbrowser	Sets the target for rpurl as the media browser pane or a secondary window.
rpvideofill-color	*color_value*	black	Sets the media playback pane color.
screensize	double|full|original	original	Sets the size at which the clip or presentation opens.
showvideocon-trolsoverlay	0|1	1	Hides the video controls overlay in the media playback pane when 0.
start	*hh:mm:ss.x*	0	Starts the clip at the specified point.
title	*text*	(none)	Specifies the clip title.

Table 5-5: Additional RealPlayer Parameters and Values

Options

There are several options to change the behavior of RealPlayer. Some of the relevant one are outlined here.

It might be desirable to reduce the size of RealPlayer, especially if your primary use is for streaming audio.

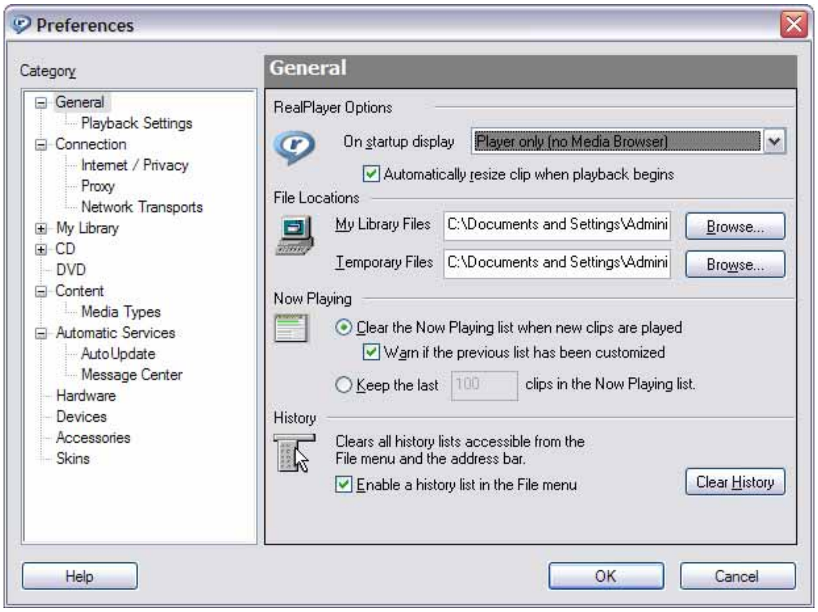
1. RealPlayer option for Player only (no Media Browser).

A) Click Tools / Preferences in the menu bar.

B) Click General.

C) Under RealPlayer Options / On startup display:
Change to Player only (no Media Browser).

D) Click OK.



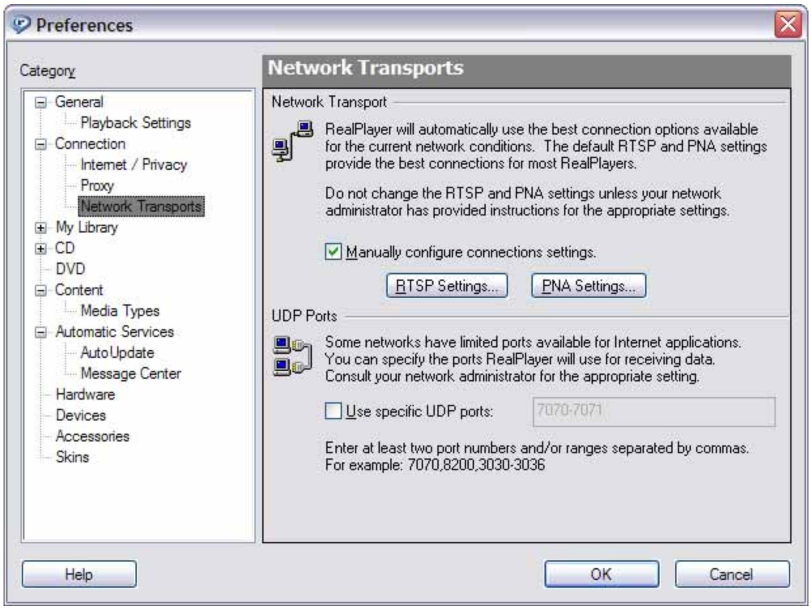
Sometimes it may be necessary to change the Network Transport settings from their default values. Certain firewalls and/or routers may not pass UDP packets, or they are unreliable and the audio stream contains excessive dropouts. In this case, change the Network Transport settings to TCP. Network overhead increases slightly when using TCP.

2. RealPlayer Network Transports

A) Click Tools / Preferences from in the menu bar.

B) Click Connection / Network Transports.

C) Under Network Transport, Check “Manually configure connections settings.”



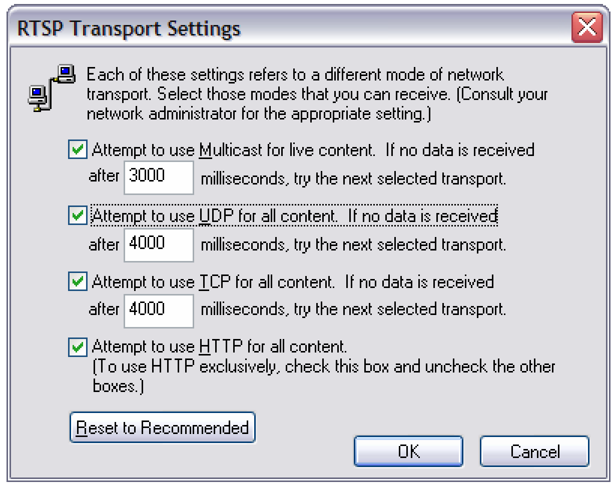
D) Click RTSP Settings.

The RTSP Transport Settings dialog box appears.

E) Uncheck "Attempt to use UDP for all content."

F) Click OK.

G) Click OK again to close Preferences.



Live/File Stream Linking in a Web Page

RealPlayer can be launched and can play streams or files directly from a streaming link in a web page. RTSP streams for RealPlayer cannot be directly referenced in HTML. Instead, do this by placing metafiles or playlist files on the web server and linking to them in the HTML code that produces the web page. Refer to the previous section for information about metafiles and playlist files, and how to create them.

This section assumes you have knowledge of HTML authoring and tags. You must have write access to the web server in order to place the necessary files there.

1. HTML HREF Tag.

A) Open a text editor program such as Notepad or an HTML editor.

B) Create the following HTML:

Example HTML file – playlist file on same server/directory:

```
<html>
<head>
<title>STREAM LINK</title>
</head>
<body>
<a href="stream.ram">STREAM LINK</a>
</body>
</html>
```

Example HTML file – playlist file on different server:

```
<html>
<head>
```

```
<title>STREAM LINK</title>
</head>
<body>
<a href="http://123.45.67.8/stream/stream.ram">STREAM LINK</a>
</body>
</html>
```

C) Save the file with a .html file extension when done.

D) The HTML file may be tested locally by double-clicking the HTML file.

A web browser will open with your stream-linked web page.

E) Clicking on the Stream Link will launch RealPlayer, connect, and play the stream.

2. Move the HTML and Playlist Files to the Web Server.

A) Using an ftp client or whatever means necessary, move the .html and .ram files to the web server.

B) Test the streaming link on the newly uploaded page to make sure all the paths and directories are correct.

Embedded Players

Not yet.

Apple Computer QuickTime 6

QuickTime is used to play RTSP/RTP streams served from an RTSP/RTP server such as Darwin Streaming Server. QuickTime has HTML embedded player capability for embedding players into web pages and/or creating custom players. QuickTime supports private authenticated streams supported by Darwin Streaming Server. At this time QuickTimePlayer supports only AAC. Unfortunately, it does not have support for HE AAC/aacPlus. We hope that Apple Computer will add this soon.

Figure 5-2: Apple QuickTime 6 GUI (shown playing a live Opticodec-PC stream)

Direct URL Entry and Play

1. **Start QuickTime.**

   A) Go to the START MENU / QUICKTIME or double-click the Taskbar Icon.

   B) Click FILE / OPEN URL in the menu bar.

   An Open URL dialog box will appear.

2. **Enter the URL of the live stream.**

   A) If the live stream is served from the server root media directory, the syntax is:

   `rtsp://ip.address.or.domain:rtsp_port/stream.sdp`

   Examples:

   `rtsp://123.45.67.8:554/live.sdp`

   `rtsp://opticodec.net:554/live.sdp`

   B) If the live stream is served from a directory within the server root media directory, the syntax is:

   `rtsp://ip.address.or.domain:rtsp_port/directory/stream.sdp`

   Examples:

   `rtsp://123.45.67.8:554/source1/live.sdp`
   `rtsp://opticodec.net:554/source2/live.sdp`

3. **Enter the URL of the file stream.**

   A) If the file stream is served from the server root media directory, the syntax is:

   ```
   rtsp://ip.address.or.domain:rtsp_port/file.mp4
   ```

   Examples:

   ```
   rtsp://123.45.67.8:554/file.mp4
   rtsp://opticodec.net:554/file.mp4
   ```

   B) If the file stream is served from a directory within the server root media directory, the syntax is:

   ```
   rtsp://ip.address.or.domain:rtsp_port/directory/file.mp4
   ```

   Examples:

   ```
   rtsp://123.45.67.8:554/source1/file.mp4
   rtsp://opticodec.net:554/source2/file.mp4
   ```

Metafile/Playlist Files

QuickTime can play live or file streams that are referenced in metafiles, playlist, media link, reference movie, or poster movie files. This is useful for creating aliases or shortcuts that let the user double-click to launch QuickTime automatically and start playing the stream, for distribution to other users, or for use as streaming links on web pages. Metafiles, playlist, media link and some reference movie files are text files that simply contain information about the stream, such as the stream URL, and depending upon the type of file, some optional information to control the player.

There are two different types of files for use with QuickTime, which use different file extensions. Both .mov and .qtl files are text files and can be used to launch QuickTime Player. Microsoft Windows systems register both of these file types upon a QuickTime installation, but only the .qtl extension is exclusive to QuickTime. Hence, using the .qtl extension is a reliable way to ensure that double-clicking the file launches QuickTime. It is highly unlikely that another program would have appropriated the .qtl file extension. The .qtl file also ensures that the stream will be open in QuickTime Player rather than the embedded QuickTime Plugin. The .qtl file is an XML based file and additional features include the ability to pass additional instructions to QuickTime Player.

1. **Create .mov File(s).**

   A) Open a text editor program such as Notepad.

   B) Create an entry containing the streaming URL.

   ```
   rtsptext rtsp://ip_address_or_domain:rtsp_port/directory/stream.sdp
   ```

   Example .mov files:

   ```
   rtsptext rtsp://123.45.67.8:554/live.sdp
   ```

   ```
   rtsptext rtsp://opticodec.net:554/source/live.sdp
   ```

C) When finished, save the file with a .mov file extension.

Double-clicking the .mov file will launch QuickTime, connect, and play the stream.

2. Create .qtl File(s).

A) Open a text editor program such as Notepad.

B) Create an entry containing the XML header information and the streaming URL.

```
<?xml version="1.0"?>
<?quicktime type="application/x-quicktime-media-link"?>
<embed src="ip_address_or_domain:rtsp_port/directory/stream.sdp" />
```

Additional QuickTime Player Parameters and Values

Include these parameters inside the <embed> tag on the third line of the .qtl file.

Parameter	Value	Default	Function
`autoplay=`	”true” “false”	(none)	Automatic play regardless of player setting.
`controller=`	”true” “false”	(true)	Player controller controls exposure.
`fullscreen=`	“normal” “double” “half” “current” “full”	(normal)	Player video screen size.
`href=`	“url”	(none)	Links player video to URL. Not available for audio only.
`loop=`	“true” “false” “palindrome”	(false)	Player file repeat.
`playeveryframe=`	“true” “false”	(true)	Plays all video frames. Mutes the audio.
qtnext*n*=	*n*”url/path” GOTO*n* T<myself>	(none)	Play a sequence of files/streams.
`quitwhendone=`	“true” “false”	(false)	Player quits when file is complete.

Table 5-6: Additional QuickTime Player Parameters and Values

Example live stream .qtl file:

```
<?xml version="1.0"?>
<?quicktime type="application/x-quicktime-media-link"?>
<embed src="rtsp://123.45.67.8:554/live.sdp" />

<?xml version="1.0"?>
```

```
<?quicktime type="application/x-quicktime-media-link"?>
<embed src="rtsp://opticodec.net:554/live.sdp" controller=”false” />
```

Example file stream .qtl file:

```
<?xml version="1.0"?>
<?quicktime type="application/x-quicktime-media-link"?>
<embed src="rtsp://123.45.67.8:554/file.mp4" />

<?xml version="1.0"?>
<?quicktime type="application/x-quicktime-media-link"?>
<embed src="rtsp://opticodec.net:554/file.mp4" controller=”false” />
```

Do not forget /> at the end of the file. It will not work without it.

C) Save the file with a .qtl file extension when done.

D) Double-clicking the .qtl file will launch QuickTime, connect and play the stream.

Options

There are several options to change the behavior of QuickTime. Some of the relevant ones are outlined here.

Figure 5-3 shows the default Streaming Transport Settings:

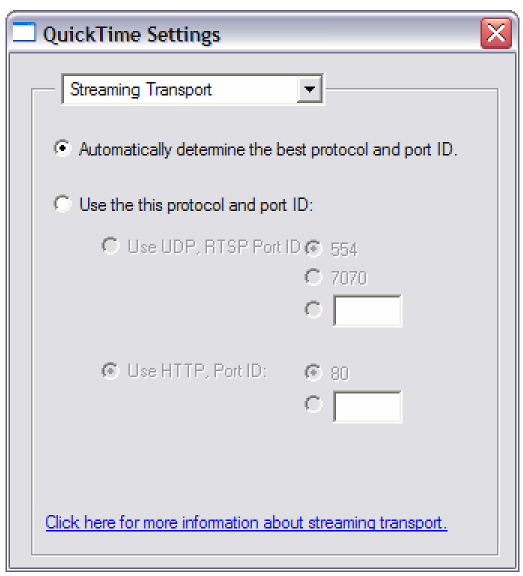


Figure 5-3: QuickTime Default Streaming Transport Settings

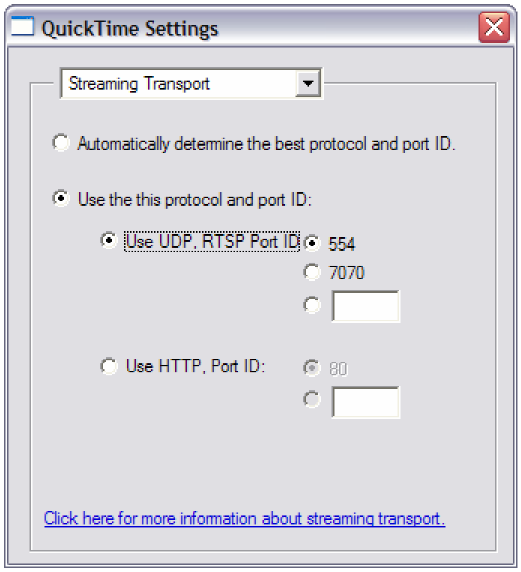


Figure 5-4: QuickTime UDP Settings

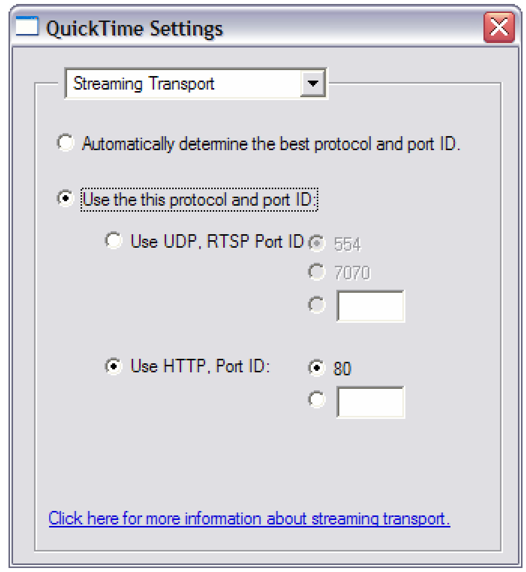


Figure 5-5: QuickTime HTTP Settings

Live/File Stream Linking in a Web Page

To play streams or files, QuickTime can be launched directly from a streaming link in a web page files. RTSP streams for QuickTime cannot be directly referenced in HTML. Instead, this is done by placing metafiles, playlist, media link, reference movie, or poster movie files on the web server and linking to them in the HTML code that produces the web page. Refer to *Metafile/Playlist Files* (starting on page 5-11) for information about these files and how to create them.

This section assumes you have knowledge of HTML authoring and tags. You must have write access to the web server in order to place the necessary files there.

1. HTML HREF Tag.

A) Open a text editor program such as Notepad or an HTML editor.

B) Create the following HTML:

Example HTML file – playlist file on same server/directory:

```
<html>
<head>
<title>STREAM LINK</title>
</head>
<body>
<a href="live.qtl">STREAM LINK</a>
</body>
</html>
```

Example HTML file – playlist file on different server:

```
<html>
<head>
```

```
<title>STREAM LINK</title>
</head>
<body>
<a href="http://123.45.67.8/stream/file.qtl">STREAM LINK</a>
</body>
</html>
```

C) When finished, save the file with a .html file extension.

You can test the HTML file locally by double-clicking it. A web browser will open with your stream-linked web page.

Clicking on the Stream Link will launch QuickTime Player, which will connect and play the stream. Any other QuickTime Player instructions described in the .qtl file will also be invoked.

2. Move the HTML and Playlist Files to the Web Server.

A) Using an ftp client or whatever means necessary, move the .html and .ram files to the web server.

B) Test the streaming link on the newly uploaded page to make sure all the paths and directories are correct.

Embedded Players

Not yet.

Nullsoft Winamp 5.05

Winamp is used to play HTTP/ICY streams and files served from an HTTP/ICY server such as SHOUTcast DNAS or Icecast2 server.

Figure 5-6: Winamp 5.05 GUI (shown playing a live Opticodec-PC stream)

Direct URL Entry and Play

1. Start Winamp.

A) Go to the START MENU / WINAMP or use the Taskbar Icon.

B) Click FILE / PLAY URL in the menu bar.

An Open URL dialog box will appear.

2. Enter the URL of the stream.

A) If the live stream is served from SHOUTcast DNAS, the syntax is:

```
http://ip.address.or.domain:port
```

example:

```
http://123.45.67.8:8000
http://opticodec.net:8000
```

SHOUTcast DNAS currently does not support content streaming of .aac files.

B) If the file or live stream is served from Icecast2, the syntax is:

For streams:

```
http://ip.address.or.domain:port/path/mountpoint.aac
```

Mountpoints may contain a directory structure:

example:

```
http://123.45.67.8:8000/path/stream.aac
http://opticodec.net:8000/stream.aac
```

For files:

`http://ip.address.or.domain:port/subdirectory/file.aac`

example:

```
http://123.45.67.8:8000/aac/file.aac
http://opticodec.net:8000/file.aac
```

- The Icecast2 webroot root directory is not specified when streaming files.
- Only the subdirectories within them are specified if used.
- Subdirectories are optional.
- Files and streams of the same name in common root locations are to be avoided. The file stream will always have priority over the live stream, i.e., a live stream with a name of `test.aac` and a file with a name of `test.aac` located in the directory specified in the root of webrootwebroot.

Metafile/Playlist Files

Winamp can play streams or files that are referenced in metafiles or playlist files. This is useful for creating aliases or shortcuts that let the user double-click to launch Winamp automatically and start playing the stream, for distribution to other users, or for use as streaming links on web pages. Metafiles, playlist, media link and some reference movie files are text files that simply contain information about the stream, such as the stream URL, and depending upon the type of file, some additional information.

There are two common types of metafiles or playlist files, .m3u, and .pls. Before either of these files can launch Winamp, they must be correctly associated with it. Most Microsoft Windows systems have .m3u associated with Windows Media Player by default. Because of this, we recommend using the .pls file, and be sure it is associated with Winamp. If you are using metafiles or playlist files to link the streams on a web page, it is a good idea to have a simple explanation about this near the link.

1. Create .m3u File(s).

A) Open a text editor program such as Notepad.

B) Create an entry containing the streaming URL.

Example .m3u file:

```
http://123.45.67.8:8000
```

C) Save the file with a .m3u file extension when done.

D) Double-clicking the .m3u file will launch Winamp, connect and play the stream.

2. Create .pls File(s).

A) Open a text editor program such as Notepad.

B) Create an entry containing these items and the streaming URL.

.pls file format:

```
[playlist]
numberofentries=<n>
File1=<uri>
Title1=<title>
Length1=<length or -1>
File2=<uri>
Title2=<title>
Length2=<length or -1>
...
Version=2
```

`[playlist]`
Signifies that this is a playlist.

`File#=`
Location of the file or stream. The # sign after "File" signifies the file or stream number. The first file in the playlist is "File1", the second is "File2," and so on. This entry can be a specific or relative path or a URL.

`Title#=`
Title to display (Optional). This is usually the title read from the file name or ID3 tags. This also can be the name of a stream. The # sign after "Title" signifies the file or stream number.

`Length#=`
Length in seconds. "–1" forces the time entry to be ignored and is used for unspecified or live streams. The "#" sign after "Length" signifies the file or stream number.

`NumberOfEntries=#`
The total number of entries in the playlist. This should match the last number on the "File#," "Title#," and "Length#" fields.

`Version=2`
This required entry at the bottom tells the player what format the PLS is. Older versions of the PLS format did not include this.

All fields are case sensitive.

Title information from .pls files will appear in the Winamp Playlist Editor. It will not appear in the main Winamp Player.

Metadata streamed from SHOUTcast DNAS or Icecast2 will take priority over Title information in the .pls file.

If SHOUTcast DNAS is used with the IntroFile feature, the Stream Name sent by Opticodec-PC Streaming Encoder (not the Title specified in the

.pls file or what may be specified in the ID3v2 tag of the file) will be displayed in Winamp while the IntroFile plays.

If a .pls file specifies a file to stream, only the filename will be displayed in Winamp, not the Title specified in the .pls file or what may be specified in the ID3v2 tag of the file.

If Playlist Shuffling is enabled in the Winamp Player, the playlist order specified in the .pls file may not necessarily play in the same order.

Example .pls files:

Single server:

```
[playlist]
numberofentries=1
File1=http://123.45.67.8:8000
Title1=Orban HiFi Internet Audio
Length1=-1
Version=2
```

Multiple servers:

Multiple servers may be listed in order to accommodate higher network traffic and/or server capacity. As one server reaches capacity, the next server in the list will attempt to deliver the stream.

Files and live streams may be referenced in the same .pls file, for example, to play a stream ID or pre-announce file before the live stream starts, however if SHOUTcast DNAS is used, it is better to use the IntroFile feature, since there will be no re-buffering between sources.

SHOUTcast and Icecast2 servers may be referenced in the same .pls file.

```
[playlist]
numberofentries=4
File1=http://stream1.opticodec.net:8000
Title1=Orban HiFi Internet Audio (Feed 1)
Length1=-1
File2=http://stream2.opticodec.net:8000
Title2=Orban HiFi Internet Audio (Feed 2)
Length2=-1
File3=http://stream3.opticodec.net:8000
Title3=Orban HiFi Internet Audio (Feed 3)
Length3=-1
File4=http://stream4.opticodec.net:8000
Title4=Orban HiFi Internet Audio (Feed 4)
Length4=-1
Version=2
```

C) Save the file with a .pls file extension when done.

D) Double-clicking the .pls file will launch Winamp, connect and play the stream.

Options

There are several options to change the behavior of Winamp. Some of the relevant one are outlined here.

Since Winamp is fully compatible with all permutations of .m3u and .pls playlist files, and their referenced files and streams, we recommend configuring Winamp to be the registered player for these file types. Both .m3u and .pls should be highlighted in Winamp Preferences / File Types. If you would like Winamp to be the registered player for other file types, highlight them as well.

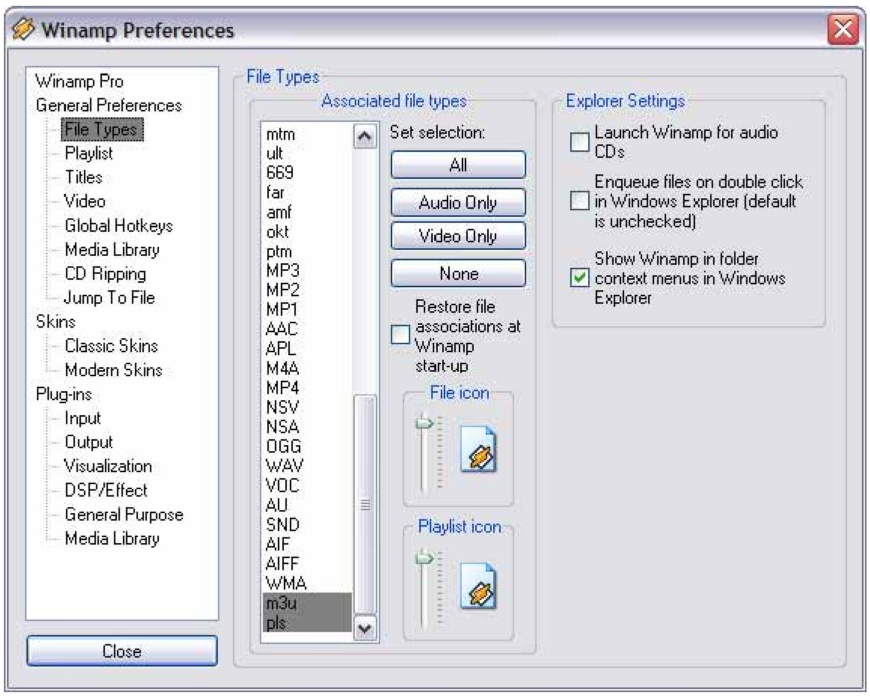


Figure 5-7: Winamp 5.05 Preferences – General Preferences / File Types

Live/File Stream Linking in a Web Page

Winamp may be launched and play streams or files directly from a streaming link in a web page. This is done by placing metafiles or playlist files on the web server and linking to them in the HTML code that produces the web page. Refer to *Metafile/Playlist Files* (page 5-17) for information about Winamp metafiles and playlist files, and how to create them.

This section assumes you have knowledge of HTML authoring and tags. You must have write access to the web server in order to place the necessary files there.

1. **HTML HREF Tag.**

   A) Open a text editor program such as Notepad or an HTML editor.

B) Create the following HTML:

Example HTML file – playlist file on same server/directory:

```
<html>
<head>
<title>STREAM LINK</title>
</head>
<body>
<a href="stream.pls">STREAM LINK</a>
</body>
</html>
```

Example HTML file – playlist file on different server:

```
<html>
<head>
<title>STREAM LINK</title>
</head>
<body>
<a href="http://123.45.67.8/stream/stream.pls">STREAM LINK</a>
</body>
</html>
```

C) Save the file with a .html file extension when done.

The HTML file can be tested locally by double-clicking the HTML file. A web browser will open with your stream-linked web page.

Clicking on the Stream Link will launch Winamp, connect, and play the stream.

2. Move the HTML and Playlist Files to the Web Server.

A) Using an ftp client or whatever means necessary, move the .html and .pls files to the web server.

B) Test the streaming link on the newly uploaded page to make sure all the paths and directories are correct.

Multiple Players on a Single Computer

It is possible to install and run *all* the players covered in this manual on the same computer. Through careful installation and configuration, these players can all co-exist without any conflicts. On Microsoft Windows computers, the main considerations are the file associations.

Section 6

Service Providers

If you do not have some or all the required resources available to you to implement your streaming application, you may need to get assistance from one or more outside service providers that specialize in streaming media and the associated technology.

Content Delivery – Hosting Services

To deliver your audio content to your intended audience on the Internet, reliable Internet connectivity is necessary. Depending upon your streaming media delivery requirements using RTSP/RTP and/or HTTP/ICY servers, regional Content Delivery Networks (CDN), Internet Service providers (ISP), and/or Internet Co-location (Colo) can provide the necessary hosting services to stream your Opticodec-PC encoded live streams and/or files to a worldwide audience on the Internet. These types of providers have services that range from Internet connectivity only, to equipment rack space for servers, to complete hosted and managed servers. To determine what services to employ, you must decide how much or how little control you want over your stream and how much you want to pay an external provider. We advise you to supply as much equipment and IT resources as possible. This will minimize your operating costs.

Content Encoding – Audio Production

If your streaming requirements call for on-demand file content delivery, Opticodec-PC AAC/HE AAC/aacPlus File Encoders provide the highest audio quality. Content encoding service providers can deliver encoded audio content for RTSP or HTTP streaming to Internet or Intranet audiences and for DVD and CD-ROM delivery from your audio files. You can commission some of these providers to do custom audio production for promos, commercials, etc.

Authoring – Multimedia Web Design

Design firms exist that create web pages for streaming content, embedded players, and complete custom player designs to your specifications. These firms provide custom authoring services, a wide variety of authoring and media integration capabilities, and workflow and production consulting with Opticodec-PC streaming compatibility.

Live Encoding – Streaming & Netcasting

Live event coverage requires audio mixing equipment, encoding computer or computers, and remote Intranet and/or Internet connectivity. Service providers should use Opticodec-PC AAC/HE AAC/aacPlus Streaming Encoders to capture and stream your events on a corporate Intranet or to the entire Internet for the highest audio quality.

Section 7

References

Information

The publications and links listed here provide more information on streaming technologies. They will give you a thorough understanding of the different streaming server technologies and topologies, as well as important information regarding interfacing to the Internet. Many of these publications and links also cover HTML and SMIL authoring to create web pages for your streaming content.

It is important to note that the Opticodec-PC AAC/HE AAC/aacPlus codec technology is new and many of these publications do not yet cover it. However, the delivery protocols are the same as those used with older codecs. As with many modern technologies, streaming is an evolving technology, so new publications and Internet resources will appear from time to time.

Books and Publications

Steven Gulie
QuickTime for the Web: For Windows and Macintosh, Third Edition (QuickTime Developer Series)
Morgan Kaufmann Publishers, Apple Computer, Inc., 2004

Jon Luini, Allen Whitman
Streaming Audio: The FezGuys' Guide
New Riders Publishing, 2002

Eyal Menin, Eyal Menin
The Streaming Media Handbook
Pearson Education, Inc, Prentice Hall PTR, 2003

Steve Mack
Streaming Media Bible
Hungary Minds, Inc., 2002

Mary Slowinski, Tim Kennedy
SMIL: Adding Multimedia to the Web
Sams Publishing, 2002

Dick C. A. Bulterman, Lloyd Rutledge
SMIL 2.0: Interactive Multimedia for Web and Mobile Devices
X.media.publishing Springer-Verlag, 2004

Gregory C. Demetriades
Streaming Media: Building and Implementing a Complete Streaming System
Wiley Publishing, Inc., 2003

Michael Topic
Streaming Media Demystified
McGraw-Hill Companies, Inc., 2002

David Austerberry
The Technology of Video and Audio Streaming
Focal Press, 2002

Joseph G. Follansbee
Get Streaming! : Quick Steps to Delivering Audio and Video Online
Focal Press, 2004

Internet Links

Apple QuickTime

Main QuickTime Page
http://www.apple.com/quicktime/

QuickTime Products
http://www.apple.com/quicktime/products/

QuickTime and QuickTime Pro
http://www.apple.com/quicktime/products/qt/

QuickTime Streaming Server
http://www.apple.com/quicktime/products/qtss/

Darwin Streaming Server
http://developer.apple.com/darwin/projects/streaming/

QuickTime Documentation
http://developer.apple.com/documentation/QuickTime/

QuickTime for the Web
http://developer.apple.com/documentation/QuickTime/QTBooks/QT4WebBook.htm

Icecast

Main Page
http://www.icecast.org/

Downloads
http://www.icecast.org/download.php

Documents
http://www.icecast.org/docs.php

RealNetworks

RealPlayer PC
http://www.real.com/

RealPlayer Macintosh
http://www.real.com/mac/?src=072604realhome_1_3_2_1_1_1

RealPlayer Linux
http://www.real.com/linux/?src=072604realhome_1_3_2_1_1_1

RealPlayer Mobile
http://www.realnetworks.com/industries/mobile/operators/products/player/index.htm
l?src=072604realhome_1_3_2_1_1_1

SHOUTcast

Main Page
http://www.shoutcast.com/

SHOUTcast Support Documentation
http://www.shoutcast.com/support/docs/

Winamp Client Player
http://www.winamp.com/

SMIL

SMIL 2.0
Interactive Multimedia for Web and Mobile Devices
http://www.xmediasmil.net/

SMIL:
Adding Multimedia to the Web
http://www.smilbook.com/

W3C Synchronized Multimedia
http://www.w3.org/AudioVideo/

The SMIL Tutorial
http://www.helio.org/products/smil/tutorial/toc.html

Learn SMIL with a SMIL
http://www.empirenet.com/%7Ejoseram/index.html

The CWI SMIL Page
http://homepages.cwi.nl/~media/SMIL/

Oratix GRiNS for SMIL
http://www.oratrix.com/GRiNS/

Fluition
http://www.fluition.com/

Adobe Systems GoLive
http://www.adobe.com/products/golive/pdfs/golive_overview.pdf

Utilities

Virtual Audio Cable
http://www.ntonyx.com

Pinguin Audio Metering
http://www.masterpinguin.de/

XMLtoRefMovie
http://www.hoddie.net/xmltorefmovie/

Server Log Analysis

enScaler
http://www.enscaler.com/index.html

enScaler MediaReports
http://www.enscaler.com/products/mediareports.html

Funnel Web Analyzer
http://www.quest.com/funnel_web/analyzer/

MRTG – Multi Router Traffic Grapher
http://mrtg.hdl.com/mrtg.htm

NetTracker Web Analytics Solutions
http://www.sane.com/

Sawmill Log Analysis
http://www.sawmill.net/

Tech System Technology – Tracking/Metering Solutions
http://www.techsystem.net

Webalizer
http://webalizer.kezako.net

RFC

The Internet Engineering Task Force RFC Repository
http://www.ietf.org/rfc.html

Zvon RFC Repository – Formatted
http://www.zvon.org/tmRFC/RFC_share/Output/index.html

Section 8

Glossary

Definitions

This glossary defines terms and spells out abbreviation used throughout this manual, as well as many other computer, network, communications, audio, video, and multimedia related terms and abbreviations. References to terms defined elsewhere in the glossary appear in italics.

The glossary is incomplete in this Preliminary Manual; undefined terms will be defined in later editions of the Manual.

1G First generation cellular mobile wireless technology that supports analog devices, mostly obsolete.

2G Second generation cellular mobile wireless circuit based technology that supports digital devices using *TDMA*, *CDMA*, and *GSM*. 2G networks provide about 9.6 Kbps throughput to handsets.

2.5G Advanced second generation cellular mobile packet based wireless technology that supports digital devices using *GSM* with *GPRS* and *EGPRS* (Edge). 2.5G networks provide about 56 Kbps throughput to handsets.

3G Third generation cellular mobile cellular wireless technology, that supports digital devices using *WCDMA CDMA2000*, and *UMTS*. 3G will allow many benefits as broadband and high speed communication representing a shift from voice-centric services to multimedia-oriented like video, voice, data, and fax services. 3G wireless technology has the ability to unify existing cellular standards such as *GSM*, *CDMA* and *TDMA*. 3G networks provide upwards of 100 – 300 Kbps, ramping to 1 – 4 mbps throughput to handsets.

3G2 file file format based on 3GPP2 standard.

3GP file file format based on 3GPP standard.

3GPP a worldwide standard for the delivery of audio over third generation (3G) cellular networks based on *MPEG-4*. 3GPP was created for use on the Global System for Mobile Communication (GSM) networks, which is the world's most popular type of 3G network. 3GPP2 was defined by a different group of telecommunications bodies called 3rd Generation Partnership Project 2 (3GPP2) for use on the second most popular type of 3G network, Code Division Multiple Access (CDMA) 2000.

3GPP and 3GPP2 formats are very similar. Both are based on the *QuickTime* file format and contain *MPEG-4* and *H.263* video, *AAC* and *AMR* audio, and 3G text. 3GPP2 adds the option to use QualComm PureVoice (QCELP) audio and Movie Fragments, a technology that allows multimedia content to be delivered incrementally over standard TCP wireless networks, providing a more immediate viewing experience for the end user.

3GPP2 See 3GPP.

802.11 IEEE evolving family of specifications for wireless local area networks (*WLAN*) using the Ethernet protocol and and Carrier Sense Multiple Access with Collision Avoidance (*CSMA/CA*) for path sharing. The original modulation used in 802.11 was phase-shift keying (*PSK*) with a maximum bandwidth of 2 Mbps. However, other schemes, such as complementary code keying (*CCK*), are used in some of the newer specifications. The newer modulation methods provide higher data speed and reduced vulnerability to interference.

802.11a one of several IEEE specifications of 802.11. 802.11a operates at radio frequencies between 5.725 GHz and 5.850 GHz and uses a modulation scheme known as orthogonal frequency-division multiplexing (*OFDM*) that is especially well suited to use in office settings. Data speeds as high as 54 Mbps are possible. There is less interference with 802.11a than with 802.11b, because 802.11a provides more available channels, and because the frequency spectrum employed by 802.11b (2.400 GHz to 2.4835 GHz) is shared with various household appliances and medical devices.

802.11b one of several IEEE specifications of 802.11. 802.11b supports bandwidth up to 11 Mbps, comparable to traditional Ethernet and operates at the same radio frequency as the original 802.11 standard, 2.4 GHz. Being an unregulated frequency, 802.11b devices can incur interference from microwave ovens, cordless phones, and other appliances using the same 2.4 GHz range. However, by installing 802.11b devices at reasonable distances from other appliances, interference can easily be avoided. 802.11b is the lowest cost in the 802.11 family. Signal range is best and is not easily obstructed, however, it has a slow maximum speed and supports fewer simultaneous users than newer implementations, such as 802.11g.

802.11g one of several IEEE specifications of 802.11. 802.11g attempts to combine the best of both 802.11a and 802.11b. 802.11g supports bandwidth up to 54 Mbps, and it uses the 2.4 Ghz frequency for greater range. Being an unregulated frequency, 802.11b devices can incur interference from microwave ovens, cordless phones, and other appliances using the same 2.4 GHz range. However, by installing 802.11b devices at reasonable distances from other appliances, interference can easily be avoided. 802.11g is backwards compatible with 802.11b, meaning that 802.11g access points will work with 802.11b wireless network adapters and vice versa. 802.11g costs more than 802.11b, however, it has a faster maximum speed, supports more simultaneous users, and the signal range is best and is not easily obstructed.

AAC (Advanced Audio Coding) AAC is an efficient lossy data compression scheme intended for audio streams and designed to replace MPEG-1 Layer 3, MP3. AAC, ISO/IEC 13818-7, is an extension of the MPEG-2 international standard, ISO/IEC 13818-3. It was further improved in MPEG-4, MPEG-4 Version 2 and MPEG-4 Version 3, ISO/IEC 14496-3. MPEG has crafted a number of AAC variants under the umbrella of MPEG-2 and MPEG-4. These variants are called profiles in MPEG-2 and object types in MPEG-4. They vary in complexity and usefulness for specific markets. This variety allows vendors to select profiles specifically optimized for their applications.

This does not stand for Apple Audio Coding, although Apple Computer brought mainstream attention to AAC by supporting it in its iPod and iTunes products. It provides better and more stable quality than MP3 at equivalent or slightly lower bitrates. Like other MPEG codecs, the AAC family of codecs can be wrapped in a variety of rights management solutions. Apple Fairplay, Real Helix, SDC, and others have all been used in commercial systems with AAC.

AAC (Advanced Audio Coding) Profiles AAC has a modular approach to encoding. Depending on the complexity of the bitstream to be encoded, the desired performance and the acceptable output, implementers may create profiles to define which of a specific set of tools to use for a particular application. The standard offers six profiles:

> **AAC Main** (Main Profile) AAC Main is AAC LC coupled with a backwards adaptive predictor. AAC Main profile is rarely used, as it requires a huge increase in complexity for very little gain in efficiency.
>
> **AAC LC** (Low Complexity Profile) is the most widely used today, but is starting to be replaced by HE AAC and aacPlus v2. It provides a good trade off between complexity, quality, and bit rate.
>
> **AAC LTP** (Long Term Prediction) AAC LTP combines AAC LC with the LTP tool. LTP has the same purpose as the backwards adaptive predictor in the MPEG-2 Main Profile and has the same limitations. Because the benefits in audio quality are still small compared to the increase in complexity, AAC LTP is rarely, if ever used and cannot be recommended for broadcasting.
>
> **AAC LD** (Low Delay) is the low delay variant of AAC. It specifically addresses the needs of two-way communication applications and sacrifices compression efficiency. Since the typical delay of AAC or AAC+SBR is usually uncritical for broadcasting systems, it cannot be recommended for broadcasting because of the loss in compression.
>
> **AAC Scalable** is for hierarchical audio coding. Used in IP based systems, it can also be useful in special broadcasting systems with robust core layers and less robust enhancement layers. However, no known broadcasting systems use AAC Scalable today because of the very special use case and the loss of compression efficiency in the higher layers.
>
> **HE AAC,** also known as aacPlus, is the combination of AAC and *SBR* Technology as specified in MPEG ISO/IEC 14496-3:2001/AMD-1: Bandwidth Extension. It is the most efficient of the named AAC profiles, so it is strongly recommended for broadcasting at bitrates of 128 K bps and below. HE AAC is not intended to have a *transparent rating*; instead, it minimizes audibly objectionable artifacts at low bitrates. Double-blind MUSHRA testing by the European Broadcasting Union rates 48 kbit/s stereo HE AAC in the “Excellent” category.

Depending on the AAC profile and the MP3 encoder, 96 kbit/s AAC or 48 kbit/s aacPlus v2 can give nearly the same or better perceptional quality as 128 kbit/s MP3. Different MP3 encoders perform differently and they produce output of sometimes wildly varying quality.

AAC file (Advanced Audio Coding file) Raw, containerless *AAC*-encoded file. It can be *MPEG-2* or *MPEG-4*, although some profiles are only available in *MPEG-4*. *MPEG-4* is the official container that supports all *AAC* profiles and versions.

aacPlus™ a trademark owned by Coding Technologies which can refer to either aacPlus v1 or aacPlus v2. aacPlus v1 is also known as high efficiency *AAC*, or *HE AAC*.

aacPlus v1 *aacPlus™*

aacPlus v2 is aacPlus v1 coupled with the MPEG Parametric Stereo (ISO/IES 14496-3:2001/AMD2: 2004) technique created by Coding Technologies and Philips. Where SBR enables audio codecs to deliver the same quality at half the bitrate, Parametric Stereo enhances the codec efficiency a second time for low-bitrate stereo signals. This combination is fully standardized by MPEG but does not have a specific profile name at this time.

access file A text file called *qtaccess* that contains information about users and groups who are authorized to access media in the directory in which the access file is stored.

administrator A user with server or directory domain administration *privileges.*

administrator computer A computer with server administrator software installed that can be used to configure and manage another computer.

AC-2 An older Dolby audio *codec*, now largely obsolete.

AC-3 Dolby's transmission audio *codec*, used in DVDs and ATSC digital television, among other applications.

ADC (analog-digital converter)

ADIF minimum structure unsync file format for storage

ADPCM (Adaptive Differential Pulse Code Modulation)

ADSL (Asymetrical Data Subscriber Line)

ADTS standard used by most *MPEG-2* MP3 files

AES/EBU

AIFF/AIFC

AMR (adaptive multirate) Speech codec used for 3G wireless devices.

AMR-WB

announced broadcast A method such as *Automatic Unicast (Announce)* enabling a broadcaster to negotiate with a server to accept a broadcast.

ARP

ARQ

ATAPI

Automatic Unicast (Announce) A method of delivering a broadcast to a streaming server in which an SDP file is automatically copied and kept current on the server. A *broadcast user* name and password must be created before starting such a broadcast.

AVI (Audio Visual Interleave) A Windows video file format.

bandwidth The capacity of a network connection, measured in *bits* or *bytes* per second, for carrying data.

BER (bit error rate)

BGP (Border Gateway Protocol)

bit A single piece of information, with a value of either 0 or 1.

bitrate The speed at which bits are transmitted on a network, usually expressed in bits per second.

bitstream

broadcast Transmitting one copy of a stream over the whole network.

broadcast user A user who has permission to broadcast to the streaming server. The broadcast user name and password are set in the General Settings pane of Streaming Server Admin and are used in conjunction with *announced broadcasts.* It is not necessary to create a broadcast user for *UDP* broadcasts.

browser plug-in Software that you attach to a browser to enable it to display specific data formats.

byte Eight *bits.*

CCITT (Comite Consultatif Internationale de Telegraphie et Telephonie) An international committee based in Geneva, Switzerland, that recommends telecommunications standards, including the audio **co**mpression/**dec**ompression standards (codecs) and the famous V. standards for modem speed and compression (V.90 and so on). Although this organization changed its name to *ITU-T* (International Telecommunications Union-Telecommunication), the old French name lives on.

CCK (Complementary Code Keying)

CDMA

CDMA2000

CDMAone

CDN (Content Delivery Network)

CELP (Code Excited Linear Prediction)

CIFS (Common Internet File System) formerly SMB

CIR (Committed Information Rate) When you order a virtual circuit for a service such as frame relay or ATM, you can specify a guaranteed data rate that you want the carrier to provide. The data rate is negotiated with the carrier as the CIR (committed information rate).

When the data rate exceeds the CIR, the network starts dropping packets, so CIR should be a balance between the minimum and maximum bandwidth requirements. You can also negotiate a burst rate that lets you exceed the CIR rate to accommodate spikes in traffic. The ability to burst depends on whether bandwidth is available. CIR may also be negotiated as variable over time, so that during busy business hours more bandwidth is available.

Basically, CIR is the throughput rate that you negotiate with a service provider, and they will usually attempt to guarantee that rate. One way the carrier guarantees CIR is by dropping non-CIR traffic.

clip

clipping

client The user-side software or computer used to display *streaming* media.

codec Any technology for **co**mpressing and **dec**ompressing data. Codecs can be implemented in software, hardware, or a combination of both.

CSMA/CA (Carrier Sense Multiple Access with Collision Avoidance)

Colo (Internet Colocation)

DAB (Digital Audio Broadcasting)

DAC (digital-analog converter)

data rate Amount of information per second. Usually measured in *bits per second*.

DAW (digital audio workstation)

DBS (Direct Broadcast Satellite)

dB (decibel)

dBfs

decibel

DHCP

diff scale

distortion

dither

DMA (Direct Memory Access)

DMZ (Demilitarized Zone) A screened-subnet firewall. The DMZ is outside the internal network, but is still secured by the firewall. Both the internal network and the external network, usually the Internet, can access computers in the DMZ, but network traffic cannot be directly transferred across the DMZ.

DNS (Domain Name Service) A service that translates host names to *IP addresses.*

DOCSIS

DRM (Digital Rights Management)

DSD (Direct Stream Digital) SACD

DSL (digital subscriber line) A broadband data transmission technology that operates over telephone lines.

DSS (Darwin Streaming Server) Apple Computer open-source version of QuickTime Streaming Server (*QTSS*).

DV (digital video) A digital tape-recording format using approximately 5:1 compression to produce Betacam quality on a very small cassette.

DVB (Digital Video Broadcasting)

DVD (Digital Versatile Disk)

DVD-Audio *DVD* using specified audio format.

DVD-Video *DVD* using specified video format.

dynamic IP address

EDGE

EIDE (Enhanced Integrated Drive Electronics)

EGPRS (Edge)

embedded system

EVDO

FED (forward error correction)

firewall Software or hardware that protects the network applications and networking stack of a computer workstation or server. IP Firewall services, which can be part of computer operating systems software, scans incoming IP *packets* and rejects or accepts these packets based on a set of filters you create.

FireWire A hardware technology for exchanging data with peripheral devices, defined by *IEEE* Standard 1394. Also called iLink.

flags

frame A single image in a *movie* or sequence of images.

frame rate In a *movie,* the number of frames per second.

FTP (File Transfer Protocol) A *protocol* that allows computers to transfer files over a network. FTP *clients* using any operating system that supports FTP can connect to a file server and download files, depending on their access *privileges.* Most Internet browsers and a number of freeware applications can be used to access an FTP server.

G.711 ITU (International Telecommunication Union) standard that specifies Pulse Code Modulation (*PCM*) of voice frequencies for a 3-KHz bandwidth at 48 kbps, 56 kbps and 64 kbps.

G.712

G.722 ITU (International Telecommunication Union) standard that specifies audio for a 7-KHz bandwidth at 48 kbps, 56 kbps and 64 kbps using Adaptive Differential Pulse Code Modulation (*ADPCM*) coding.

G.723 ITU (International Telecommunication Union) standard that specifies audio transmitted at 5.3 kbps to 6.3 kbps, close to the quality of speech during a conventional phone call.

G.726

G.728 ITU (International Telecommunication Union) standard that specifies audio for a 3 KHz bandwidth at 16 kbps, using Low-Delay Code Excited Linear Prediction (*LD-CELP*).

G.729 ITU (International Telecommunication Union) standard that specifies audio for a toll-quality 8 kbps speech encoder, using linear prediction analysis-by-synthesis coder.

group ID

GPRS

GSM

H.261 ITU (International Telecommunication Union) increasingly obsolete standard video *codec* specifically designed for transmission over *ISDN* lines in that data rates are multiples of 64 Kbit/s. The standard supports CIF and QCIF video frames at resolutions of 352x288 and 176x144 respectively. The coding algorithm is a hybrid of inter-picture prediction, transform coding, and motion compensation. The datarate of the coding algorithm was designed to be able to be set to between 40 Kbits/s and 2 Mbits/s. The inter-picture prediction removes temporal redundancy. Transform coding removes the spatial redundancy. Motion vectors are used to help the *codec* compensate for motion. To remove any further redundancy in the transmitted *bitstream*, variable length coding is used.

H.262

H.263 ITU (International Telecommunication Union) standard video *codec* designed as a low bitrate encoding solution for videoconferencing. It was first designed to be utilized in H.323 based systems, but now is finding use in *RTSP* (streaming media) and *SIP* (Internet conferencing) solutions as well.

H.263 was based on *H.261*, the previous *ITU*-T standard for video compression, and provides a suitable replacement for it at all bitrates. It has been superseded by H.263v2 (a.k.a. H.263+ or H.263 1998).

H.264 MPEG-4 part 10

H.323

HD (High Definition)

HDMI (High Definition Multimedia Interface) A high-bandwidth digital interface that allows high quality video and audio data to be output at high rates of speed, up to 5 GBps. Since it does not perform unnecessary D/A and A/D conversions, the signal remains in the digital domain. HDMI accommodates all of the current ATSC digital television formats, plus it supports up to eight channels of audio.

HE AAC See AAC (Advanced Audio Coding) Profiles.

headroom

hinting Hinting creates a *track* for each streamable media track in the file that tells QuickTime/Darwin Streaming Server how and when to deliver each frame of media. The hinting process performs the required calculations in advance, allowing *QTSS* to serve up a larger number of streams. Hinting also allows new codecs to be used without the need to upgrade the server.

HTML (Hypertext Markup Language) The code inserted in a file to be displayed on a web browser page. The markup tells the web browser how to display a web page's words and images for the user.

HTTP (Hypertext Transfer Protocol) An application *protocol* using TCP that defines the set of rules for linking and exchanging files on a network.

IANA (Internet Assigned Numbers Authority)

Icecast open-source streaming media server project based on HTTP/ICY protocols.

ICMP

ICY (I Can Yell) HTTP SHOUTcast/Icecast streaming protocol.

ID3 tag

ID3v1

ID3v2

IDE (Integrated Drive Electronics)

IEEE (Institute of Electrical and Electronics Engineers) An organization dedicated to promoting standards in computing and electrical engineering.

IGMP (Internet Group Management Protocol)

IMA (Interactive Multimedia Association) audio compression algorithm supported by QuickTime with minimal quality degradation and a compression factor of 4 to 1.

IMD(intermodulation distortion)

Instant On An advance in Apple's patent-pending Skip Protection technology that-dramatically reduces buffer, or wait, time for an instantaneous viewing experience with *streaming* video on a broadband connection.

Internet

IP (Internet Protocol) A connectionless protocol used to transmit *packets* of data from one machine to another. *TCP* and *UDP* use *IP* for their host-to-host data communications.

IP address A unique numeric address that identifies a computer on the Internet.

IP gateway

IP subnet A portion of an *IP* network, which may be a physically independent network segment, which shares a network address with other portions of the network and is identified by a subnet number.

IPV4

IPV6

ISDN (It Still Does Nothing)

ISMA compliant

ISP (Internet service provider) A business that sells Internet access and often provides web hosting for ecommerce applications as well as mail services.

ITU (International Telecomunications Union) Headquartered in Geneva, Switzerland is an international organization within the United Nations System where governments and the private sector coordinate global telecom networks and services.

ITU-D ITU – Telecommunication Development Sector) One of three sectors within the *ITU*, which has well-established programs of activities to facilitate connectivity and access, foster policy, regulatory and network readiness, expand human capacity through training programs, formulate financing strategies and e-enable enterprises in developing countries.

ITU-R (ITU – Radiocommunication Sector) One of three sectors within the *ITU*, which plays a vital role in the management of the radio-frequency spectrum and satellite orbits. These finite natural resources are increasingly in demand from a large number of services such as fixed, mobile, broadcasting, amateur, space research, meteorology, global positioning systems, environmental monitoring and, those communication services that ensure safety of life at sea and in the skies.

ITU-T (ITU – Telecommunication Standardization Sector) One of three sectors within the *ITU* created on 1 March 1993, replacing the former International Telegraph and Telephone Consultative Committee (*CCITT*), whose origins go back to 1865. The public and the private sectors cooperate within *ITU-T* for the development of standards that benefit telecommunication users worldwide.

jitter

JavaScript A scripting language used to add interactivity to web pages.

key frame A sample in a sequence of temporally compressed samples that does not rely on other samples in the sequence for any of its information. Key frames are placed into *temporally compressed* sequences at a frequency that is determined by the *key frame rate.*

key frame rate The frequency with which *key frames* are placed into *temporally compressed* data sequences.

LAN (local area network) A network maintained within a facility, as opposed to a WAN (wide area network) that links geographically separated facilities.

LATM (Low-overhead MPEG-4 Audio Transport Multiplex)

layer A mechanism for prioritizing the *tracks* in a movie or the overlapping of sprites. When it plays a movie, *QuickTime* displays the movie's images according to their layer—images with lower layer numbers are displayed on top; images with higher layer numbers may be obscured by images with lower layer numbers.

LD-CELP (Low-Delay Code Excited Linear Prediction)

lossless codec

lossy codec A method where compressing a file and then decompressing it retrieves a file that may well be different to the original, but is "close enough" to be useful in some way. It is used a lot on the Internet and especially in streaming media and telephony applications

LRMP

M3U file An audio metafile that is created using a text editor and saved to a web server. The file directs a user's web browser to an MP3 playlist residing on the same web server and opens the user's MP3 player.

M4A file

M4P file

Mac OS X The latest version of the Apple operating system, which combines the reliability of UNIX with the ease of use of Macintosh.

Mac OS X Server An industrial-strength server platform that supports Mac, Windows, UNIX, and Linux clients out of the box and provides a suite of scalable workgroup and network services plus advanced remote management tools.

Manual Unicast A method for transmitting a live stream to a single *QuickTime* Player client or to a computer running *QTSS or DSS.* An SDP file is usually created by the broadcaster application and then must be manually sent to the viewer or streaming server.

Mbone (Multicast Backbone) A virtual network for real-time *streaming* over the Internet.

MIB/MIB-II (Management Information Base) for network management of *TCP/IP*-based networks.

MIDI (Musical Instrument Digital Interface) A standard format for sending instructions to a musical synthesizer.

MIME type (Multipurpose Internet Mail Extension)

modifier track A *track* in a movie that modifies the data or presentation of other tracks. For example, a *tween* track is a modifier track.

mount point A string used to identify a live stream, which can be a relayed movie stream, a nonrelayed movie stream, or an MP3 stream. Mount points that describe live movie streams always end with an .sdp extension.

MOV file The Apple QuickTime *movie* file extension used to name both movie redirect files and actual *QuickTime* media files.

movie A structure of time-based data that is managed by *QuickTime.* A *QuickTime* movie may contain sound, video, animation, or a combination of data types. A *QuickTime* movie contains one or more *tracks;* each track represents a single data stream in the movie.

MP2 file A file format for *MPEG-1 Layer 2*.

MP3 file A file format for *MPEG-1 Layer 3*.

MP4 file A file format for *MPEG-4*.

MPEG-1 Layer 2 A popular professional format for compressing and storing and delivering audio that can achieve *transparent ratings* at higher bitrates. It is used for broadcast digital playout systems, broadcast links, satellite audio, background music delivery, LAN/WAN streaming, and DAB.

MPEG-1 Layer 3 A popular format for compressing audio that does not quite achieve *transparent ratings* at any bitrate. It is used for temporary broadcast links, consumer audio applications, inefficient Internet streaming, and many other places it really shouldn't. *AAC* is designed to replace *MPEG-1 Layer 3* (commonly called *MP3*)..

MPEG-1 A popular format for compressing video and audio used for V-CD. Due to the poor quality of its video, it was not accepted in many parts of the world. It was replaced by MPEG-2.

MPEG-2 A popular format for compressing video and audio used for DVD-Video, DBS, and DVB.

MPEG-4 An *ISO* standard based on the QuickTime file format that defines multimedia file and compression formats. This is a container format.

MPEG-4 ALS (MPEG-4 Audio Lossless Coding) As an extension to MPEG-4 audio, the amendment, ISO/IEC 14496-3:2001/AMD 4, Audio Lossless Coding (ALS) defines methods for *lossless* coding. The basic technology for MPEG-4 ALS was developed by the NUe Group (Fachgebiet Nachrichtenübertragung) at Technical University of Berlin.

multicast An efficient, one-to-many form of *streaming.* Users can join or leave a multicast but cannot otherwise interact with it.

multihomed A server with multiple *IP addresses.*

NAT (Network Address Translation) A technique sometimes used so that multiple computers can share a single *IP address.*

netcast A broadcast of live video or audio on a network or the Internet.

NIC (network interface card)

NNTP

NTU (network termination unit)

ODFM (Orthogonal Frequency-Division Multiplexing)

open source A term for the cooperative development of software by the Internet community. The basic principle is to involve as many people as possible in writing and debugging code by publishing the source code and encouraging the formation of a large community of developers who will submit modifications and enhancements.

packet A unit of data information consisting of header, information, error detection, and trailer records. *QTSS/DSS* uses *TCP, UDP,* and *IP* packets to communicate with streaming clients.

payload

PDC

PDU (Protocol Data Unit) Information that is delivered as a unit among peer entities of a network and that may contain control information, address information, or data. In layered systems, a unit of data that is specified in a protocol of a given layer and that consists of protocol-control information of the given layer and possibly user data of that layer.

PHS

pixel A single dot in a graphic image with a given color and brightness value.

playlist A set of media files in the *QTSS* or DSS media folder specified to play one after the other or in random sequence.

POP3 (Post Office Protocol)

port A sort of virtual mail slot. A server uses port numbers to determine which application should receive data *packets. Firewalls* use port numbers to determine whether data packets are allowed to traverse a local network. “Port” usually refers to either a *TCP* or *UDP* port.

PPPoE (Point-to-Point Protocol over Ethernet)

PPTP (Point-to-Point Tunneling Protocol) A protocol for a Virtual Private Network (VPN) developed by a consortium including Microsoft and is used for establishing VPN (Virtual Private Network) tunnels across the Internet. This allows remote users to securely and inexpensively access their corporate network from anywhere on the Internet. PPTP uses a client-server model for establishing VPN connections. Most Microsoft operating systems ship with a PPTP client, so there is no need to purchase third-party client software. PPTP has the additional advantage over other VPN technologies of being easy to setup.

PPP (Point-to-Point Protocol)

privileges Settings that define the kind of access users have to shared items. You can assign four types of privileges to a share point, folder, or file: read and write, read only, write only, and none (no access).

progressive download Audio or video data that is pushed via *HTTP* to the client player. The audio or video can be listened or viewed by the user as it is being transferred. This is not a form of media *streaming.*

protocol A set of rules that determines how data is sent back and forth between two applications.

proxy server A server that sits between a client and server and negotiates communication between those two hosts. The client and server only communicate with the proxy server and never interact with each other. There are proxy applications for many network protocols, including *HTTP* (for web traffic) and *RTSP* (for streaming traffic).

PSK (Phase Shift Keying)

QCELP (QualComm Code Excited Linear Preditive Coding) Speech codec used in *3GPP2 CCK* **(Complementary Code Keying)**

CDMA devices.

QoS (Quality of Service)

qtaccess The name of the plain text *access file* that contains information about users and groups who are authorized to view media in the directory in which the access file is stored.

qtgroups

qtusers

QTSS (QuickTime Streaming Server) A technology that lets you deliver media over the Internet in real time.

QuickTime A set of Macintosh system extensions or a Windows dynamic-link library that supports the composition and playing of *movies.*

QuickTime Player An application, included with the *QuickTime* system software, that plays QuickTime *movies.*

QuickTime Pro A version of *QuickTime Player* with advanced features, primarily the addition of editing capabilities.

RAID (Redundant Array of Independent Disks) A hard disk array that either increases the speed of disk input-output or mirrors the data for redundancy, or provides both of these features. Users may access the RAID as if it were one drive, although it may be divided into multiple partitions.

RAM file Real Audio metafile format used for linking streams to be played using RealPlayer.

RCA plug

RDT

Reliable UDP

reference movie A .mov file created using a utility program like MakeRefMovie, available at no cost from Apple for Macintosh and Windows. The file contains the location of a streaming media file and can also contain the locations of multiple streaming files. A reference file linked from a web page, for example, can direct a client player to the on-demand presentation encoded for its particular connection speed.

reflected stream A file or live broadcast delivered as a *unicast* stream. *movie* and MP4 *QTSS*/DSS *playlists* also generate reflected streams.

relayed stream A stream that is passed from one server to one or more other servers. Relays can also be used to generate a *multicast* stream.

RFC

RIFF

RLE (run length encoded)

RTP (Real-Time Transport Protocol) A network-transport protocol used for transmitting streaming real-time multimedia content over *multicast* or *unicast* network services, usually used with *RTSP* and *RTCP*.

RTCP (Real Time Control Protocol)

RTSP (Real Time Streaming Protocol) A *protocol* for controlling a stream of real-time multimedia content. Sources of data can include both live feeds and stored clips.

SACD

sample rate The number of samples per second used for audio. Higher sample rates yield higher quality audio than lower sample rates.

SAP (Session Announcement Protocol) A *protocol* used to announce Internet multicast conferencing sessions. A conference is announced by periodically multicasting a *UDP* announcement packet to a multicast address and port. Because SAP is designed for multicast, it is suitable for setting up conference calls, not one-on-one *IP* telephone calls.

SBR (Spectral Band Replication) is a technology that reduces the bitrate necessary for audio codecs to provide high (although not transparent) subjective audio quality. It transmits only a lower frequency band by means of its base codec, while the receiver synthesizes higher frequencies from the lower frequency band by a process similar to that used in harmonic-generating “exciters.” A low-bitrate control signal is transmitted to the receiver. This signal contains hints for the receiver that help it conform the generated high frequency signal to the original high frequency signal present at the transmitter.

scope also known as TTL

SCSI (Small Computer System Interface) a parallel interface standard used by computers for attaching peripheral devices, mainly storage.

SDP (Session Description Protocol) A text file used with *QTSS*/Darwin Streaming Server that provides information about the format, timing, and authorship of a live *streaming* broadcast and gives the user’s computer instructions for tuning in.

SHOUTcast™

signal-to-noise ratio

SIP (Session Initiation Protocol) An Internet protocol that provides simple application layer signaling for setting up, maintaining, and terminating multimedia sessions such as voice calls, videoconferences, and even instant messaging sessions. SIP performs many of the functions of the ITU H.323 multimedia conferencing standard, which was largely specified by the telecoms. SIP provides a more-scalable, higher-performance, and more-efficient calling model. Because it is designed on the Internet model, it is inherently distributed and supports the development of telephony applications on Internet systems.

SMB (Server Message Block) Renamed CIFS

SMIL

SMTP (Simple Mail Transport Protocol) A *protocol* for sending email.

sprite An animated image that is managed by *QuickTime.* A sprite is defined once and is then animated by commands that change its position or appearance.

SNMP

SNR (signal-to-noise ratio)

S/PDIF

SPI (Stateful Packet Inspection)

SSL (Secure Sockets Layer) An Internet *protocol* that allows you to send encrypted, authenticated information across the Internet.

static IP address An *IP address* that is assigned to a computer or device once and is never changed.

streaming Delivery of audio or video data over a network in real-time, as a stream of *packets* instead of a single file download.

TCP (Transmission Control Protocol) A method used along with the Internet Protocol (IP) to send data in the form of message units between computers over the Internet. *IP* takes care of handling the actual delivery of the data, and TCP takes care of keeping track of the individual units of data (called *packets*) into which a message is divided for efficient routing through the Internet.

TDMA

temporal compression Image compression that is performed between *frames* in a sequence. This compression technique takes advantage of redundancy between adjacent frames in a sequence to reduce the amount of data that is required to accurately represent each frame in the sequence. Sequences that have been temporally compressed typically contain *key frames* at regular intervals.

TFTP

track A *QuickTime* data structure that represents a single data stream in a QuickTime *movie*. A movie may contain one or more *tracks.* Each track is independent of other tracks in the movie and represents its own data stream.

transparent rating When a codec's output cannot be distinguished from its input in a statistically significant manner by a panel of expert listeners or viewers in biascontrolled (usually double-blind) tests using critical program material, the codec is said to be *transparent*.

TTL (time-to-live) *A multicast* broadcast has a TTL value that is set by the user. It specifies the number of routers the stream will pass through before it stops propagating over the network.

tween track A *track* that modifies the display of other tracks.

UDP (User Datagram Protocol) A data transport *protocol* that does not support retransmission of lost *packets,* sometimes used instead of *TCPIP.*

UNIX

UMTS (Universal Mobile Telecommunications System)

unicast The one-to-one form of *streaming.* If *RTSP* is provided, the user can move freely from point to point in an on-demand *movie.*

URL (Universal Resource Locator) A uniform way of specifying locations on the Internet or a local file system.

USB

USB2

VBR (variable bitrate) A method of compressing data that takes advantage of changes in the media's *data rate.*

WAN (Wide Area Network)

WAP

WAV file A Microsoft Windows format for sound files.

WCDMA

webcast More properly termed *netcast*, a broadcast of live video or audio on a network or the Internet.

WEP (Wired Equivalent Privacy)

WiFi

Windows CE

Windows 2000

Windows 2000 Server

Windows 2003 Server

Windows XP

WLAN (Wireless Local Area Network)

WMA (Windows Media Audio) Microsoft proprietary audio codec.

WMT (Windows Media Technologies) Microsoft proprietary technology for delivery and storage of multimedia based content.

WMV (Windows Media Video) Microsoft proprietary video codec.

WPA (Wi-Fi Protected Access)

WWW (World Wide Web) Referred to as "the Web," is a collection of pages on the *Internet* located on servers all around the world that can be read and interacted with by computer.

XLR plug A three-pin audio connector that can be used with three-wire balanced cables, which cause electromagnetic interference to be canceled out.

XML An extensible markup language, similar to *HTML* but more formal and more flexible.