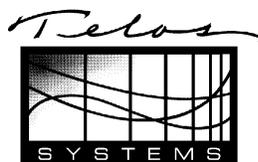


Telos Systems



Audioactive Professional Hardware MPEG Realtime Encoder

User's Manual
Manual Version 3.0.e



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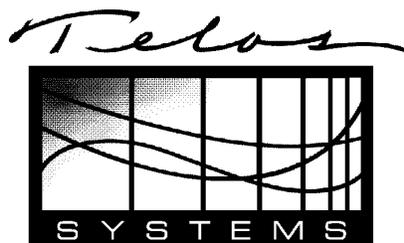
The address is: support@audioactive.com.

- **Via World Wide Web.**

The Audioactive Web site has a variety of information. The URL is: www.audioactive.com. The general Telos site is: www.telos-systems.com.

Feedback

We welcome feedback on any aspect of the Audioactive Internet Audio Suite or this manual. In the past, many good ideas from users have made their way into software revisions or new products. Please contact us with your comments.



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Updates

A continuous program of improvement is underway. Please be sure to send in the registration form to be notified of new software releases.

Notice

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Cautions and Electrical Warnings

CAUTION: The installation and servicing instructions in this manual are for use by qualified personnel only. To avoid electric shock, do not perform any servicing other than that contained in the operating instructions unless you are qualified to do so. Refer all servicing to qualified personnel.

WARNING: To reduce the risk of electrical shock, do not expose this product to rain or moisture. Do not shower with the unit.



This symbol, wherever it appears, alerts you to the presence of uninsulated, dangerous voltage inside the enclosure – voltage which may be sufficient to constitute a risk of shock.



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



USA CLASS A COMPUTING DEVICE INFORMATION TO USER.

WARNING: This equipment generates, uses, and can radiate radio-frequency energy. If it is not installed and used as directed by this manual, it may cause interference to radio communication. This equipment complies with the limits for a Class A computing device, as specified by FCC Rules, Part 15, Subpart J, which are designed to provide reasonable protection against such interference when this type of equipment is operated in a commercial environment. Operation of this equipment in a residential area is likely to cause interference. If it does, the user will be required to eliminate the interference at the user's expense. **NOTE:** Objectionable interference to TV or radio reception can occur if other devices are connected to this device without the use of shielded interconnect cables. FCC rules require the use of only shielded cables.



CANADA WARNING: "This digital apparatus does not exceed the Class A limits for radio noise emissions set out in the Radio Interference Regulations of the Canadian Department of Communications." "Le present appareil numerique n'emet pas de bruits radioelectriques depassant les limites applicables aux appareils numeriques (de les Class A) prescrites dans le Reglement sur le brouillage radioelectrique edicte par le ministere des Communications du Canada."

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Introduction

Welcome to the next frontier! At Telos, we have never been content to rest in the past. While We have been famous for making leading-edge audio and broadcasting equipment, this product represents something that We feel strongly will allow you to become an important part of the future.

In the real world of the Internet, billions of variables are interacting continuously – enough to fill the Milky Way. While it would be presumptuous of us to make any kind of resolute predictions, We do have some notions about how audio on the net will play out:

- Despite competition from other distribution systems, audio on the Internet will continue to grow because it offers unlimited potential for content choice.
- Standards-based systems will prevail. As with traditional broadcasting, transmitting and receiving will not be accomplished with a closed, proprietary system. As real-time Internet media become mainstream, standards will permit listeners with players in various styles and from various sources to listen to content created using tools from any number of manufacturers.

Telos is committed to the development of technologies and tools for Internet audio transmission which offer you the ability to work in a fashion similar to the traditional broadcast paradigm - where the use of open standards allow a wide range of encoding and decoding products.

The Audioactive system is a suite of tools which allow you to deliver audio content to users connected to the Internet for real-time listening of live and stored audio streams.

Telos' solution is optimized to provide the best-possible audio quality within the bandwidth constraints imposed by the limitations of Internet capacity while also offering the flexibility to fully exploit the higher bitrate quality possible for Intranet applications.

Audioactive NetSuite Components

The Audioactive MPEG Realtime Encoder is bundled as a complete streaming solution. The included Audioactive NetSuite CD-ROM features the Audioactive Multimedia Driver, Audioactive MP3 Production Studio and Microsoft Media Technologies, a full-featured server platform. The audio streams are decoded and played by the client using the ubiquitous Windows Media player. The Audioactive Player is also included for playback of pure MP3 streams.

Hardware Audioactive MPEG Realtime Encoder

This unit is the heart of our professional solution for Internet and Intranet audio encoding. The encoding process is not constrained by the limitations of a general-purpose PC platform. A multiple DSP-chip compression engine performs state-of-the-art enhanced MP3 bitrate reduction, the powerful audio method endorsed by the competitive MPEG standards process, to ensure superior sound quality. A pro-grade 20-bit analog-to-digital converter ensures the best possible quality at the conversion stage. (Also to be referred to as Audioactive Realtime Encoder.)

The Audioactive MPEG Realtime Encoder is a professional rack-mounted hardware unit. It receives audio via balanced analog XLR connectors or digital AES/EBU input. The hardware unit then encodes the audio input into a compressed MP3 bitstream using a sophisticated multiple-DSP processing engine. Audio input sampling rate is selectable. The audio bitrate is adjustable from 8kbit/s up to 128kbit/s. The bitstream is sent to both its asynchronous RS-232 output and 10BaseT Ethernet port. The encoder software settings can be configured through the front panel, or through a terminal connected to its RS-232 control port or via a Telnet network session.

Each Internet encoder generates a single bitrate audio stream; multiple units may be used to serve streams to users at multiple rates. The produced bitstreams are compatible with Microsoft Media Technologies and the Windows Media Player.

Microsoft Media Technologies

Microsoft Media Technologies Server platform is optimized for delivery of media content via the Internet and Intranets. (This suite of products from Microsoft is also often referred to as Window Media, and NetShow.)

The professional Audioactive MPEG Realtime Encoder is used as the audio compression front-end to maintain broadcast and high-quality audio production standards. The hardware Audioactive Encoder provides a dedicated, multi DSP audio compression engine that encodes audio to high-quality MPEG Layer-3. The encoded audio is delivered via TCP/IP or an RS-232 serial connection to a Window Media Server.

The Windows Media Server software provides replication of the stream to multiple listeners, status and statistics reporting, and control capability.

The Microsoft/Telos compatibility is possible because standards based MPEG Layer-3 (MP3) is the underlying audio compression technology.

The ubiquitous Windows Media Player will be distributed with all Microsoft operating systems and Microsoft Internet Explorer. Players are freely available for Windows 3.1, 95, 98, and NT, as well as Macintosh computers and UNIX.

Taking you into the future of Webcasting today....

We understand your need to reach large audiences without the steep price tag. The Telos/Microsoft package supports IP Multicast, the next-generation Internet technology that permits very large numbers of listeners to be served economically. IP Multicast capability will eventually move audio programming on the Internet and on corporate Intranets to the next level. The high volume listener potential enabled by the Microsoft platform, combined with Audioactive's high fidelity audio compression creates a complete professional webcasting solution.

Audioactive Multimedia Driver

Telos provides the Audioactive Multimedia Driver to send the encoded audio data received from the Audioactive MPEG Realtime Encoder to the Windows Media Server. After you install the driver on your machine, the Audioactive Realtime Encoder appears as a sound device which provides MPEG Layer-3 output.

It would also be possible to run another Stream Server on the same Windows NT machine that hosts Windows Media, with the audio bitstream being "paralleled" in software.

Audioactive MP3 Production Studio Encoder

The Audioactive MP3 Production Studio encoder is a fast, high-quality audio solution for compressing audio files. This advanced software-only encoder converts standard audio files in Microsoft .wav format to compressed MPEG-2 Layer-3 .mp3, .asf, .swa, or compressed .wav files. Once the audio files are encoded, they may be used for local playback, or they may be stored on any web server for download or real-time streaming. To listen, our free Audioactive player or any of the free and shareware players that support MP3 files can be used.

Bitrates from 8Kbps to 256Kbps are supported, with stereo mode possible for 20Kbps and greater.

The Audioactive MP3 Production Studio ships with the NetSuite CD-ROM in the 30-day trial mode. To activate this program, an activation key is provided on the packaging. Available for PCs with Windows 95, 98 or NT operating systems.

Audioactive Player

The Audioactive Player is intended to be used as a “helper app” for web browsers, but may be used stand-alone. The Audioactive Player supports only pure MP3 streaming and will not accept .asf streams generated by the NetShow Server. This player is included as part of the Audioactive NetSuite CD-ROM for pure MP3 streams that can be generated by hardware Audioactive MPEG Realtime Encoder and the Audioactive MP3 Production Studio.

Audioactive freeware player for Windows 3.x/95/NT and Power Macintosh is freely available for download at www.audioactive.com

Audioactive Recorder

The Audioactive Recorder is an application to be used in conjunction with the Audioactive MPEG Realtime Encoder for audio logging and archiving. It allows the user to save an MP3 stream coming from an Audioactive Realtime Encoder through its Ethernet port and save it on hard disk. Different output formats are supported: MP3, ASF, SWA and compressed WAV. This application can be automated using scripts and Windows scheduler.

Audioactive Recorder requires a PC with Windows 95, 98 or NT operating system.

Important Terms

In order to improve clarity, We will use important terms consistently throughout the documentation. Here is a description of some of them:

System

A system consists of one or more Internet Encoders, Primary Server (optional), Replication Servers (optional), Players, and network connectivity.

Program

One audio source at a single bitrate. Each program requires its own Internet Encoder. A system may support any number of programs. Normally, a single server may support as many as eight.

Audio Bitstream (or Encoded Audio Bitstream)

This is the encoded (compressed) audio output from the hardware Audioactive Realtime Audio Encoder.

Bitrate

The audio stream bitrate is expressed in bits/second (bps) or thousands of bits/second (Kbps). The bitrate is set at the Realtime Encoder and is then passed transparently through the server system to the player. A bitrate is chosen for each program based upon the connection rate of the intended listener and the quality target. A higher bitrate provides better audio fidelity but requires more Internet bandwidth. Normally, a program intended for listeners using 28.8Kbps modem. Internet connections would be transmitted at 16 to 24Kbps, for example, because this bitrate will pass through the modem channel.

MPEG-2 Layer-3 (MP3)

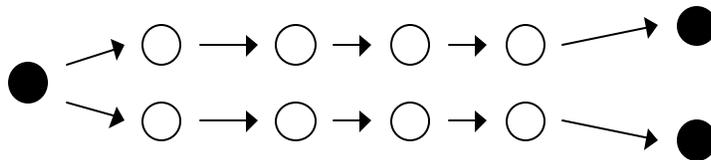
This is a high-performance audio coding method chosen by the MPEG standards body. It is optimal for music and mixed music/speech audio at Internet bitrates, from 16 to 128Kbps. At modem rates, it sounds like AM radio; at 128Kbps, it is comparable CD quality. Also known as 'MP3' among its many devotees on the Net.

ASF

Microsoft Media Technologies and the Microsoft Media Player supports the Active Streaming Format. (ASF) This format is a powerful, extensible, synchronized multimedia file format. It expands upon existing formats and incorporates new media types, improved editability, media stream prioritization, and enhanced indexing. It is designed to support the streaming of multimedia over a network from a media server or from a web server as well as for local playback, all over a wide range of bandwidth and usage scenarios.

Unicast Stream

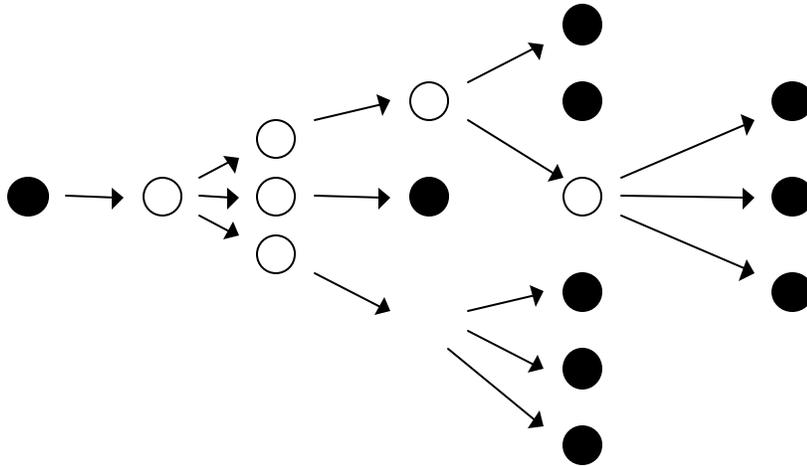
This is the bitstream which is output from the server and connects to the listener. There is one of these for each active listener.



The network has to support bandwidth all the way from the source to each listener. This is the primary method now used for audio transmission on the Internet.

Multicast Stream

This is the next generation Internet technology that permits very large numbers of listener to be served economically. IP Multicast achieves this goal by performing the splitting and replication of streams within the Internet itself, thus reducing the bandwidth and server resources required. IP Multicast capability will move audio programming on the Internet, and on the corporate Intranets, to the next level.



Multicast streams permits larger numbers of listeners without consuming network bandwidth for the full path from source to listener, since the intermediate nodes are able to replicate the stream. This option is ideal for sending a program across a corporate network since now most routers are multicast enabled, allowing a virtually unlimited number of listeners. This mode can be enabled using the Stream Server function of the Audioactive Realtime Encoder or through the appropriate option of Microsoft Windows Media Server.

IP address

Each device connected to a network needs an ID number called an IP address. In order to hook-up the Audioactive Hardware Encoder to your local network through its 10Base-T Ethernet port, you will need to ask your network administrator to assign it a fixed IP address and associated Netmask. In addition, if you want to be able to control or send a stream over the Internet you will need to enter your Internet gateway IP address.

TCP

Standard protocol for communication amongst computers across networks, TCP is used for a wide range of applications on the Internet including the popular Web and FTP applications. Each TCP connection enables a dialog between network devices allowing repetitions in case of lost of information. TCP is only used in unicast applications. The Audioactive Realtime Encoder is fully compliant with the TCP protocol.

AATCP

AATCP is a server proprietary protocol. This protocol allows Telos products and drivers to connect and control the Realtime Encoder over a network connection.

HTTP

This is the standard protocol used for web client/server applications. The Audioactive Hardware Encoder integrates a mini web server tailored for audio streaming that can be accessed using the Audioactive Player “Open Location” option. UDP

UDP is another computer network protocol used to send information between connected devices in a one-way fashion. Unlike TCP, this protocol offers no guaranty that the recipient has indeed received the information from the sender in its integrity. This mode, however, allows multicast applications (see above). The Audioactive Realtime Encoder supports UDP protocol under its Stream Sender function.

Global Thinking: Systems and Strategies

This section will describe the interrelationship of the various components and discuss issues of concern to those who will be designing a system.

Internet Bandwidth Requirements

An important concern is the Internet connection that will be used to serve the Unicast Streams. Since each listener consumes his own Internet bandwidth, the amount of bandwidth available will determine the number of listeners who may be simultaneously connected.

The formula for estimating the bandwidth requirement is:

Required Bandwidth = Number of Unicast Listeners x (encoded audio stream bitrate x 1.1)

The following table shows estimated numbers of listeners for each Internet connection style, assuming a 16Kbps encoded audio bitstream rate:

Internet Connection	Bandwidth	Listeners
ISDN Basic Rate	128Kbps	8
T1	1.5Mbps	90
Ethernet LAN	10Mbps x .7	350
T3	45Mbps	2,700
100 Base-T/FDDI LAN	100Mbps x .7	4,000

Of course, you may not wish to allocate all of your available bandwidth to audio streaming. For this purpose, the Windows Media Technologies Server permits a System Administrator to set a listener limit for each program. This function may also be used to allocate bandwidth among the various programs which may be being served.

Encoders

One hardware Audioactive MPEG Realtime Encoder is required for each program or audio stream you wish to transmit. Each Encoder can work alone as a network server and/or it can send its stream to a primary server (see below). See Appendix B for information regarding stand-alone operations.

Primary Server

The Primary Server in this manual will be referred to as the bundled the primary Windows Media Technologies Server to which the Internet Encoders can be connected. The encoder/server connection can be via a TCP/IP 10BaseT network or a distance-limited RS-232 serial connection. You may also use the Real Audio G2 Server as the primary server either stand alone or simultaneously. **Using the Real Audio G2 Server will require the MP3 plug-in available through Digital Bitcasting.**

Listeners may connect directly to this server, and/or this server may be used to provide Unicast Streams to Replication Servers, which would then provide them to listeners.

Replication Servers

While a complete system may consist of just one encoder and one server, it is possible to expand a system to include an unlimited number of additional servers. The purpose would be to accomplish any or all of the following:

- To serve a larger number of listeners than would be possible with a single server
- To locate the server closer (either geographically or “network-wise”) to the intended listeners
- To locate the server where there is more Internet bandwidth capacity available
- To offer maximum audience reach via both Microsoft Windows Media and RealNetworks G2 Servers

A Replication Server uses the normal Media Technologies or RealNetworks software, with appropriate options set to enable the replication function. It works by accepting a Unicast Stream input from the Primary Server, replicating it and distributing them to listeners.

See Media Technologies or RealNetworks documentation to configure a Replication Server.

Audioactive MPEG Realtime Encoder



This section describes the Audioactive Encoder used as part of a complete system for encoding and distributing audio content via the Internet or private networks.

The Audioactive Realtime Encoder is a professional rack-mounted hardware unit. It receives audio input through balanced XLR connectors and converts it into the digital domain using a very high-quality 20-bit analog-to-digital converter. The Audioactive Realtime Encoder then encodes the input into a compressed MPEG Layer-3 bitstream using a sophisticated multiple-DSP processing engine. The bitstream output is sent to a PC through either a TCP/IP 10BaseT network, or an asynchronous RS-232 connection.

The Realtime Encoder may be used alone to send unicast or multicast MP3 encoded bitstreams throughout a network. It can also integrate with Microsoft Media Technologies or the Digital Bitcasting plug-in for RealNetworks G2 Server to form a complete Internet “broadcasting” system. The Microsoft Windows Media Technologies software is described in separate document and kept updated directly on the Microsoft website. Please refer to the Microsoft on-line documentation to learn about the “big-picture” systems level and for other integration information.

The encoder compresses audio using the MPEG 2.5 Layer-3 international ISO standard.. It includes the latest enhancements and extensions for best audio quality at the bitrates required for Internet applications as well as near-CD quality for intranets/extranets contributions.

The encoder settings can be configured using its front panel control knob and LCD, or with a terminal on a dedicated RS-232 serial connection. Each unit can also be controlled remotely via a Telnet connection over a TCP/IP network.

The user selectable encoding options include:

- bitrate
- sampling frequency
- mono or stereo mode
- network configuration

The bitrate for Layer-3 encoded audio can be one of 8, 12, 14, 16, 18, 20, 24, 32, 40, 48, 56, 64, 96, 112 or 128Kbps.

Depending on the bitrate, the sampling frequency can be one of 11 kHz, 12kHz, 16 kHz, 22 kHz, 24 kHz, 32 kHz, 44kHz, 44 kHz, or 48 kHz. Stereo mode is supported for bitrates of 20Kbps and higher.

Connecting the Encoder: The Rear Panel

Network 10 Base-T Jack

It is possible to control the unit and receive the compressed audio bitstream via 10-BaseT Ethernet.

RS232 Audio Output

This is the serial output compressed audio bitstream. Baud rate may be set to either 115 or 230Kbps. It is connected via a female DB-9 DTE connection. Pin 2 is transmit, pin 3 is receive, and pin 5 is ground. Protocol is 8N1.

The MPEG standard bitstream is delivered to the serial port asynchronously. The bitstream is divided into eight-bit segments and sent to the output as each new byte is available. No particular alignment of serial bytes to MPEG frames is guaranteed. This output is always active independently of the network configuration.

RS232 Control

This is used to make parameter changes. It is a female DB-25 DTE connection at 19.2Kbps 8N1. Pin 2 is transmit, pin 3 is receive and pin 7 ground.

Both serial ports are configured for DTE (*Data Terminal Equipment*) as opposed to DCE (*Data Communication Equipment*) operation – meaning that it looks like a computer, not a modem. Therefore, a custom or null-modem cable is required to connect to a computer, while a standard cable is required to interface to a modem.

AES/EBU Input

This female XLR is a Digital AES/EBU input. The Realtime Encoder can accept arbitrary sample rate input.

Analog Audio Input

These are the left and right analog audio inputs. Connectors are female XLRs. Pin 1 is GND, pin 2 is positive phase audio, and pin 3 is negative phase audio. Nominal level is +4dBu with front panel set to 0dB. Peak level accepted is +18dBu.

The left channel input should be used for monaural operation.

Power Connection/Switch

Accepts input of 90-240 VAC. There is an internal 2A/250V fuse.

Operating the Encoder

Starting-up

When you first turn on the Audioactive Encoder, you will see a message on the front panel indicating that the system is loading software into its Digital Signal Processors. After this is done, the unit will be ready to go. All operating modes are stored in non-volatile memory, so values will be retained from the last time the unit was powered.

After the DSPs are loaded, you will see a menu with five options: Audio Input, Encoder Options, Output, System Options, and Network. Below the menu, you will see the current encoder settings, along with network connection.

Level Meters

The LED level bars represent the audio level after encoding has taken place. The LED's just to the right of the level meters indicate clipping of the audio input. If these flash often or stay lit, you should adjust the input lower.

(Due to the nature of the compression techniques, the levels after encoding in stereo mode are lower than ones levels after encoding in moon mode. This is normal.)

Control Knob

Rotate to change menu position or options and then push to select.

Choosing Modes

The front-panel configuration menus include help and status information on the current settings of the encoder.

The configuration options also support an "auto" mode for the sampling frequency and number of channels. With the sampling frequency and number of channels set to auto, the user only needs to choose a bitrate for the encoded audio. The encoder will then choose the best-sounding combination of sampling frequency and stereo/mono modes for that bitrate.

Status Bar

The first part of the status bar tells you what menu or submenu you are in. The top-level menu displays the "Audioactive" text logo.

Menu: Audio Input

Source

Choose analog or digital AES/EBU input. Analog is the default setting.

Level adjust

Adjusts the input level. Range is from -4dB to +15dB. 0dB is default. The level adjust only applies when the input source is set to analog.

Menu: Encoder Options

Bitrate

Selects the bitrate of the compressed audio output stream. Choose from 8, 12, 14, 16, 18, 20, 24, 32, 40, 48, 56, 64, 80, 96, 112, and 128Kbps. 64Kbps is the default.

The bitrate determines how much data will flow for each second of audio: this ultimately determines the audio quality. The higher the bitrate, the better the audio sounds. At 128 Kbit/sec, the audio has near-CD quality. However, higher rates require more transmission bandwidth. Thus, the bitrate also limits how the encoded audio can be streamed, since faster rates can exceed the capability of a modem connection.

This table shows the relationship between bitrate, compression ratio, subjective audio quality and the type of connection required for streaming applications. For example, audio to be streamed over a 28.8Kbps modem should be coded to a bitrate of 20Kbits/second or less.

Bitrate (Kbits/Sec)	Mode	Streaming Application	Bandwidth (kHz)	Subjective Audio Quality
128Kbps	Stereo	Intranets	15-20kHz	Near-CD quality
96Kbps	Stereo	128Kbps ISDN, High-speed LAN	15kHz	FM quality
56Kbps	Stereo	64Kbps ISDN High-speed LAN	12kHz	Near-FM quality
20Kbps	Stereo/Mono	28.8/33.3/56K Modems	4.7kHz	“Boombox” quality
16Kbps	Mono	28.8/33.3/56K Modems	4.1kHz	AM quality
12Kbps	Mono	14.4 Modems	3.8kHz	SW quality

Mode

Stereo, Mono or Auto. We suggest Auto, which will select the mode you have determined is appropriate for the chosen bitrate.

Samplerate

The samplerate is the rate at which the encoder’s a/d converter will sample the audio. This is also the rate at which a decoder’s hardware must be set. (With the Audioactive player, the samplerate is automatically detected and the computer’s sound card properly adjusted.)

Choose 8, 11.025, 12, 16, 22.050, 24, 32, 44.1 or 48 kHz or Auto. Again, we suggest Auto, which will choose the optimal samplerate and offers the best audio quality for the chosen bitrate. The bitrates 11.025, 22.050 and 44.1 are generally preferred because some Soundblaster cards will not accommodate other rates.

Menu: Output

Baudrate

Sets the baudrate for the RS-232 audio bitstream output. Choices are 115 and 230Kbps. Generally, it is better to choose the lowest rate that will support your chosen bitrate, as it put less demand on the server. Bitrates above 80Kbps require 230Kbps baudrate.

Menu: System Options

Contrast

Adjusts the contrast of the LCD display.

Backlight

Adjusts the backlight brightness of the LCD display.

Set Defaults

Restores all encoder settings to their default states. This option should only be used in extreme conditions to completely reconfigure the Realtime Encoder. All previous configuration information, password and, network settings will be lost.

A series of warning dialogs are displayed confirming that you wish to continue with this operation.

About

Gives information on version numbers of the firmware.

Menu: Network Configuration Menu

To use 10BaseT support, you will need to configure the unit before you can use it on the network. To do this, you will need to specify an IP address and network mask. Typically, this information will be provided to you by your network system administrator.

To configure your network settings, select the “Network” menu option on the main menu. You will see a list of the following options:

IP Address

This specifies the IP address to use for your unit. Your system administrator will typically give you an IP address to use for your unit, and you must set this before the unit can be used on the network.

The current IP address setting is displayed in brackets to the right of the IP address menu option. To set the IP address, click the control knob when the IP address option is highlighted. An entry dialog will pop up in front of the menu, allowing you to change the current IP address.

When the entry dialog cursor is flashing, you can select one of the IP address numbers to change by moving the control knob left or right. You can select either the OK or Cancel options by moving the control knob past the right side of the IP address. When the cursor is over a number on the entry screen, click the control knob to change that number's value. After the control knob is clicked, you will notice that the cursor stops flashing, and that moving the control knob left or right changes the value at that cursor position. Click the control knob again to move to another number, or to select OK/Cancel.

When you have entered your IP address, select the OK button and press the control knob. If at any time you want to exit the entry screen, you can instead select the Cancel button and press the control knob, which will exit the entry screen and return you to the network configuration menu.

A warning dialog is displayed if the unit doesn't have 10BT support and the IP address menu option is selected.

Net Mask

This allows you to change the net mask to work with the network to which your unit is connected. Your system administrator can tell you what the net mask is for your network, but it typically is a number that looks something like 255.255.255.0. You must set this value before the unit can be used on the network.

To change the network mask, click the control knob when the net mask option is selected. An entry dialog will pop up in front of the menu, allowing you to change the current net mask setting. See the IP address section for a description of how to enter information in the entry dialog.

(A warning dialog is displayed if an earlier unit doesn't have 10BT support and the Net Mask menu option is selected.)

TCP Port

The TCP port specifies what port number a client will connect when requesting coded audio data from the unit. The default is 1024.

Selecting the TCP Port option brings up an entry dialog that allows you to change the port setting. For protocols like HTTP, you may want to use the default HTTP port 80.

Protocol

When a client connects to the unit over the network, the protocol determines how the unit communicates with the network client. In the current version HTTP, AATCP, and Raw TCP protocols are supported.

1. When HTTP is selected, the unit acts as a web server for real-time audio, and can deliver audio streams to clients such as the Audioactive Windows or Macintosh player, or one of the many other shareware players that support network play options. The URL for the real-time audio stream will look like "http://<hostname>/13audio.mp3", where <hostname> is either the hostname or IP address of the unit. You can also select "<http://<hostname>/13audio.m3u>" to download a playlist file from the unit, which will point to the real-time audio stream URL.
2. The Raw TCP option specifies that no server to client negotiation protocol is used, and that coded audio is sent to a client as soon as a TCP connection on the appropriate TCP port is made.
3. The AATCP protocol is available for use with other Telos products, and allows such features as remote configuration of the unit and password restricted access for client connections.

When using the unit in conjunction with the Audioactive Multimedia driver, select the Raw TCP option. When using in a standalone operation in conjunction with the Audioactive Player, select the HTTP option (see Appendix B).

Disconnect

The disconnect option allows you to close any network connection currently active. When the Disconnect menu option is selected, a warning dialog is displayed asking if you wish to close the current network connection. Select OK and click the control knob to close the connection, or Cancel if you wish to leave it active.

Note that if a connection is currently active, the "net" icon will be displayed in the lower right hand corner of the display.

Menu: Stream Sender

This menu allows setup of the built-in UDP stream server. Notice that although it uses the same IP addresses configuration as the TCP server, its operations are independent and thus it can work simultaneously while TCP connections are active.

With Stream Sender you can send a stream to several listeners either based on their IP address (unicast mode) or as a program for multiple listeners (multicast mode).

Unlike TCP, a UDP server sends a stream regardless of the request of a player. Thus only the server, in this case the Audioactive Encoder has total control over the network stream.

Add Host

Specify the IP address and port number for the recipient of the UDP stream. The port specifies what port number a client will connect when requesting coded audio data from the unit. The default is 1024. Depending upon the type of IP address entered, Stream Sender will act as a unicast or as a multicast server.

The Audioactive Encoder keeps a list of Host IP addresses and ports in its non-volatile memory.

Selecting a multicast IP address (in the range 224.0.0.1 through 239.225.225.225) brings up an entry dialog that allows you to change the Multicast Time To Live (TTL) setting. (See Appendix B for recommendations regarding the choice of multicast IP address and proper TTL setting).

NOTE: In order to use the multicast mode, a multicast enabled router should be available on the network in order to replicate the stream.

Delete Host

Use this option to remove an IP & port settings from the Host list.

Start Sender

Select the Host you wish to send the UDP stream to. After confirmation, the Encoder will start sending its stream to the specified Host. The network logo should appear on the lower right corner of the LCD screen. If this is not the case, there might be a network problem, make sure that the Host is available on the network. Note that the Encoder will keep sending the stream to the specified Host until it is stopped (see below Stop Sender option).

The Audioactive Encoder memorizes the active Stream Senders and will attempt to restart them after a recycling power.

Stop Sender

Use this option to stop sending a UDP stream to a specific Host.

Disconnect All

Use this option to stop all the currently active Stream Senders.

Using Telnet/RS232 remote control

Telnet/RS232 remote control

The Audioactive Realtime Encoder can be controlled over a network using Telnet, or through a direct serial connection using a terminal program. A maximum of two Telnet sessions can be active at any given time.

Using Telnet

To use Telnet, you have to first configure the network settings of the Realtime Encoder for your network. When you connect to the Realtime Encoder over Telnet, the first thing displayed is the “login:” prompt, followed by a cursor. To log in, enter the Telnet password for your Realtime Encoder. As you type in your password the characters will not be displayed on the screen.

The default Telnet password is “telos”. You can change the Telnet password from the front panel by going to the Telnet submenu in the network menu.

NOTE: it is strongly recommended that you change the Telnet password if your Realtime Encoder is going to be on a publicly accessible network. See your network administrator if you have any questions.

Once you have logged in, you will see the following:

```
Welcome to Audioactive Realtime Encoder version X.
```

```
Last login from xxx.xxx.xxx.xxx
```

```
>>_
```

You are now ready to type in commands. To get a list of all the commands available, type “?” or “help” at the prompt. To get help with any specific command, type the command name followed by the “?” character. You can type “exit” at any time to end your Telnet session.

You can also change the Telnet password directly through Telnet. For example, if your new password is “newpass”, you can type the following:

```
telnetpwd telos newpass newpass
```

The telnetpwd command takes three arguments: the current password, followed by the new password, and the new password again to verify that you have typed it in correctly. Your new password will take effect for all subsequent Telnet connections to the Realtime Encoder, and will be saved even after the unit is powered down.

Using a direct serial connection

If you wish to control the Realtime Encoder from a direct serial connection, you can connect your computer to the DB-25 “remote control” port on the back of the Realtime Encoder.

NOTE: Depending upon your computer’s serial port and the type of serial cable you are using, you may need to use a NULL modem adapter. For specific information on the remote control port pinout, see section <21>.

Set your terminal program for 19200 bits per second, 8 data bits, No parity, 1 stop bit, and no flow control. When you hit enter, you should see the “>>” prompt. If you don’t, verify that your terminal program is set up properly and that the serial cable is properly connected to your computer and the Realtime Encoder.

Once you are connected, you are ready to log in. At the command prompt, type “login telos”. The default password for a direct serial connection is telos. You should see the following:

```
>> login telos
```

```
Logged on to level 3.
```

NOTE: Even though the default password for a serial connection is the same as the default password for Telnet, the two passwords are independent. You can change one without affecting the other.

To change the serial login password, use the “passwd” command. For example, if your new serial login password is “serpass”, you can type the following:

```
passwd telos serpass serpass
```

The passwd command takes three arguments: the current password, followed by the new password, and the new password again to verify that you have typed it in correctly. To save the password so that it will still take effect after the unit is powered down, type “passwdsave” at the command prompt.

Remote Control Commands

Once you have logged in through Telnet or a direct serial connection, you can type “?” to see all the commands available to you. A list of commands is displayed, organized by what part of the Realtime Encoder functionality the affect. You should see something like the following

```
>> ?
```

Command System

? echo help login
logout passwd passwdsave

Netcoder Application

version

Network

connections netinfo ping readmac
setgateway setip setmask setport

Operating System

date time

Stream sender

senderadd senderdel senders senderstart
senderstop

System

encoding para paras paravalues

Telnet

exit telnetpwd

>>

To get help on a specific command, type the command name followed by the “?” character, or equivalently “help” followed by the command name. For example, if you wanted to find out what the “version” command does, you could type “version ?” and you would see the following:

```
>> version ?  
Syntax: no args  
Desc: displays software version information
```

Or similarly:

```
>> help version  
Syntax: no args  
Desc: displays software version information
```

This tells us that the version command takes no arguments, and that it should tell us about the

version of software running on the Realtime Encoder. If you type “version”, you should see something like the following:

```
>> version
Audioactive Realtime Audio Encoder
Copyright 1998 TLS Corp. All rights reserved.
Application Version: 4.10
LoadsetVersion: v3.10
Front Panel Version: 2.00
>> _
```

Each of the remote control commands is described below.

Command

```
>> ??
Syntax: [<topic>][<topic>]
Desc: get a list of commands when no topic,
or additional info about a given command
```

```
Command>> echo ?
Syntax: [on | char | line | prompt | noprompt | off]
Desc: change char echo and enable/disable prompt
```

Command

```
>> help ?
Syntax: [<topic>]
Desc: get a list of commands when no topic,
or additional info about a given command
Command
```

```
>> login ?
Syntax: [password]
Desc: Login to higher command level
```

The only command level available to the user is command level 3. Command level 1 is the default for serial connections when no one is logged in. Note that the Telnet password does not apply to the “login” command.

Command

>> logout ?

Syntax: no args

Desc: logout of current command level

Logs the user out of the current command level. When using a direct serial connection, this command can be used to return the command system to level1 for security reasons.

Command

>> passwd ?

Syntax: <oldpasswd> <newpasswd> <newpasswd>

Desc: change password of current level

Note that the passwd command is separate from the telnetpwd command, and does not affect the Telnet password.

Command

>> version ?

Syntax: no args

Desc: displays software version information

Command

>> connections ?

Syntax: no args

Desc: Show information on active network connections

Displays detailed information on any client connections to the Realtime Encoder, active stream senders, and Telnet connections. For example, typing "connections" would show information that looks something like the following:

>> connections

```
<type> <address>:<port>      <connection duration>
-----
client 207.20.69.125           2 days 4 hours 1 minute
client 192.224.22.223          3 hours 25 minutes
client 207.20.69.122           12 hours 4 minutes
UDP to 192.206.40.135:1024     1 minute
```

Total of 4 connection(s)

Telnet:

1 Telnet session(s) active

last login from 207.20.69.125

>>

This tells us that there are 3 active client connections, one sender active, and a single Telnet session currently active.

Command

>> netinfo ?

Syntax: no args

Desc: Show information on network configuration

For example, typing “netinfo” would show information similar to the following:

>> netinfo

Network settings:

IP address: 127.20.69.125

Net mask: 255.255.255.0

Gateway: 127.20.69.1

Server port: 80

Command

>> ping ?

Syntax: <x.x.x.x>

Desc: Ping host at IP address

The ping command can be used for debugging network problems, and will indicate if the Realtime Encoder can reach the given host over the network.

WARNING: Don't use the ping command when there are audio connections to the network server, as it may cause interruptions in the audio.

Command

>> readmac ?

Syntax: no args

Desc: Read the mac address from Ethernet card EPROM

For example, typing “readmac” from the command line would show something like the following:

>> readmac

Current mac address: 00 50 C2 00 00 01

Command

>> setgateway ?

Syntax: <x.x.x.x>

Desc: Set or change the default IP gateway

Command

>> setip ?

Syntax: <x.x.x.x>

Desc: Set or change the IP address

Command

>> setmask ?

Syntax: <x.x.x.x>

Desc: Set or change the IP network mask

Command

>> setport ?

Syntax: <port>

Desc: Set the TCP port used by the audio data connection server

Command

>> date ?

Syntax: <month> <day> [<year>]

Desc: set system date or get if no arguments

Note that the Audioactive Realtime Encoder does not have a real-time clock, so if the unit is powered down, any date or time information set using these commands will be lost.

>> date

System time: 1-1-1970 00:27:28

Command

>> time ?

Syntax: <24hour> <minute> [<second>]

Desc: set system time or get if no arguments

Note that the Audioactive Realtime Encoder does not have a real-time clock, so if the unit is powered down, any date or time information set using these commands will be lost.

>> time

System time: 1-1-1970 00:27:28

Command

>> senderadd ?

Syntax: <x.x.x.x> <port>
Desc: Add an stream sender host

Command

>> senderdel ?
Syntax: <x.x.x.x> <port>
Desc: Remove a stream sender host

Command

>> senders ?
Syntax: no args
Desc: Show all defined stream sender hosts

For example, typing “senders” at the command prompt should show something like the following:

```
>>senders
Network senders:
  inactive: 202.10.30.2:1024
  sending: 127.20.69.125:1024
```

Command

>> senderstart ?
Syntax: <x.x.x.x> <port>
Desc: Initiate a stream sender connection

Command

>> senderstop ?
Syntax: <x.x.x.x> <port>
Desc: Stop a stream sender connection

Command

>> encoding ?
Syntax: [<bitrate> <freq|auto> <auto|mono|stereo|val>]
Desc: Shows actual encoder settings, or specify args to set

For example, to set the encoder to 80 kbps, auto sampling frequency, and mono, you would type “encoding 80 auto mono”, and should see something like the following:

```
>> encoding 80 auto mono
Looking for 80Kbps, auto samplerate, mono...
```

Setting encoder mode, b s m = 80 255 3
Found bitrate 80kbps, samplerate AUTO (44kHz), mode (L+R)/2 Mono
saving to NVRAM...
Bitrate 80 kbps, samplerate 44 kHz, mode Mono

Similarly, if you just wanted to know what sampling frequency was selected for the “auto” setting, you could type “encoding” and would see something like the following:

```
>> encoding  
Bitrate 80 kbps, samplerate 44 kHz, mode Mono
```

Command

```
>> para ?
```

Syntax: <para index> [<value index>]

Desc: Changes or displays a system parameter setting

Each of the system parameters has a corresponding index. To see a list of all system parameters and their associated index values, use the “paras” command.

When the para command is used with one argument, it displays the current setting of the parameter specified. For example, if you wanted to see what the input source was set to, typing “paras” shows us that the input source parameter has index 4, so you type “para 4” and should see something like the following:

```
>>para 4  
Input Source set to Analog (idx: 1)
```

Which shows us that the input source is set to analog (index 1).

To change the value of a parameter, you would use the para command with two arguments. The first argument is the index of the parameter you want to change, and the second argument is the value you want to change it to. The value has an index value as well, and to see what these index values are for a parameter you can use the “paravalues” command.

For example, if you want to see what the other selections for the input source parameter are, you would type “paravalues 4”

```
>>paravalues 4  
* 1) Analog    2) Digital
```

So, if you wanted to change the input source from analog to digital, you would type “para 4 2”. The first argument is the parameter index (4 for input source), and the second argument is the index for the value you want (2 for digital):

```
>>para 4 2  
saving to NVRAM...  
Input Source set to Digital (idx: 2)
```

Note that the para command saves any changes that you make so that the next time the unit is powered down, your changes will still take affect.

Command

>> paras ?

Syntax: no args

Desc: Shows all system parameters and their current settings

For example, typing the "paras" command will display a list of parameters by parameter index:

>> paras

- | | |
|-------------------------|---------------------------|
| 1) Bitrate [64 kbps] | 2) Samplerate [AUTO] |
| 3) Mode [AUTO] | 4) Input Source [Analog] |
| 5) Input Level [0db] | 6) Contrast [60] |
| 7) Backlight [80] | 8) Data RS232 [230 kBaud] |
| 9) Use DHCP [No] | 10) Protocol [HTTP] |
| 11) Server active [Yes] | |

Command

>> paravalues ?

Syntax: <para index>

Desc: lists all possible values for the specified parameter

For example, to see the possible values for the "Input Level" parameter (parameter index 5), you would see the following:

>> paravalues 5

- | | | | |
|-----------|-----------|-----------|-----------|
| 1) -4db | 2) -3db | 3) -2db | 4) -1db |
| * 5) 0db | 6) +1db | 7) +2db | 8) +3db |
| 9) +4db | 10) +5db | 11) +6db | 12) +7db |
| 13) +8db | 14) +9db | 15) +10db | 16) +11db |
| 17) +12db | 18) +13db | 19) +14db | 20) +15db |

Command

>> exit ?

Syntax: no args

Desc: Logout from telnet session

Command

>> telnetpwd ?

Syntax: <oldpasswd> <newpasswd> <newpasswd>

Desc: change telnet login password

Realtime Encoder Using Audioactive Realtime Encoder with Microsoft Windows Media Technologies

Telos provides the Audioactive Multimedia Driver to send the encoded audio data received from the Audioactive Realtime Encoder to the Windows Media Server. After you install the driver on your machine, the Audioactive Realtime Encoder appears as a sound device which provides MPEG Layer-3 output.

Microsoft uses Active Streaming Format (ASF) to store and transport multimedia content. Media Technologies server comes with the ASF real-time Encoder tool. This tool encapsulates the MPEG Layer-3 audio stream from the Audioactive Realtime Encoder using ASF and sends the ASF stream to the Windows Media Server. Microsoft Media Players can then connect to the Windows Media Server and are able to listen to the stream. The diagram below illustrates this process.

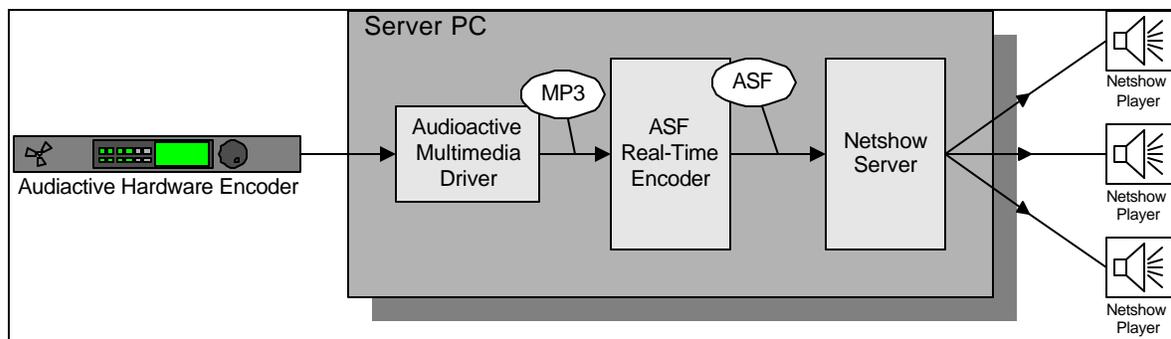


Figure 1: Using Audioactive with the Windows Media Server

Audioactive Multimedia Driver

Audioactive Realtime Encoders may be connected to the server using TCP/IP anywhere on the network or directly to a PC through an RS-232 serial connection. The Audioactive Multimedia Driver kit contains two helper applications that allow you to specify the encoders you wish to use as audio sources for the multimedia driver. When the Audioactive Multimedia Driver connects to the hardware unit over a network, the RAW TCP/IP protocol should be specified on the Realtime Encoder.

Audioactive Port Manager allows you to configure the communication ports used by Audioactive Realtime Encoders connected directly to your PC via RS-232. Audioactive TCP Manager allows you to configure the TCP/IP parameters for encoders connected through TCP/IP.

Driver Installation:

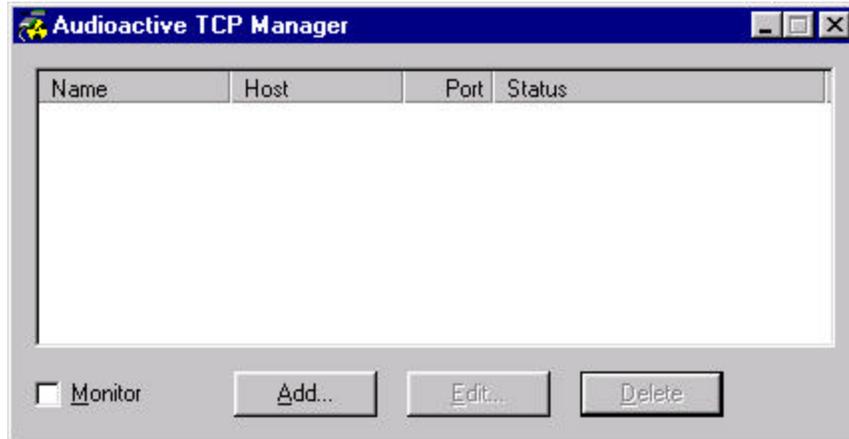
The Audioactive Multimedia Driver is shipped on CD-ROM. To install the driver on your PC run the SETUP program. The setup process will copy the driver files to your PC and update the registry information for you.

NOTE: Make sure that any Audioactive Realtime Encoders connected directly to the PC via RS-232 are turned off while Windows NT boots up. After the driver software is installed, the Encoders may be left on at all times.

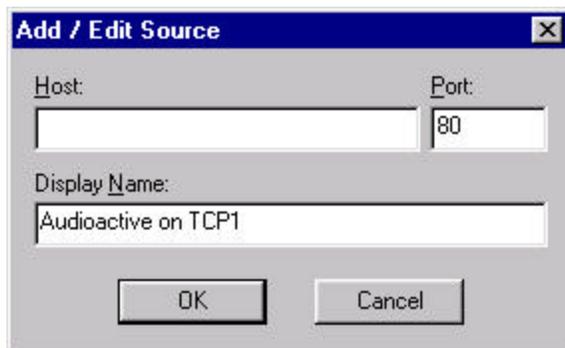
Audioactive TCP Manager Application

You should use Audioactive TCP Manager to manage Audioactive Realtime Encoders that are connected to a network. Use Audioactive Port Manager (see page 33) to configure encoders connected to your PC through an RS-232 communication port.

1. Start Audioactive TCP Manager. Since no encoders were defined yet, the display is blank.



2. Click the Add button.



3. Enter the IP address of the encoder and the port number assigned. The default value is 80.
4. Enter a name for this encoder. This name will appear when you later select the audio source for Windows Media Technologies' ASF Real-Time Encoder so you may want to choose a descriptive name.
5. Click the OK button. You should see the information you entered displayed by the program. You should be able to see the encoder settings in the status column.

You can listen to the audio provided by the encoder by selecting the desired encoder and clicking the Monitor checkbox.

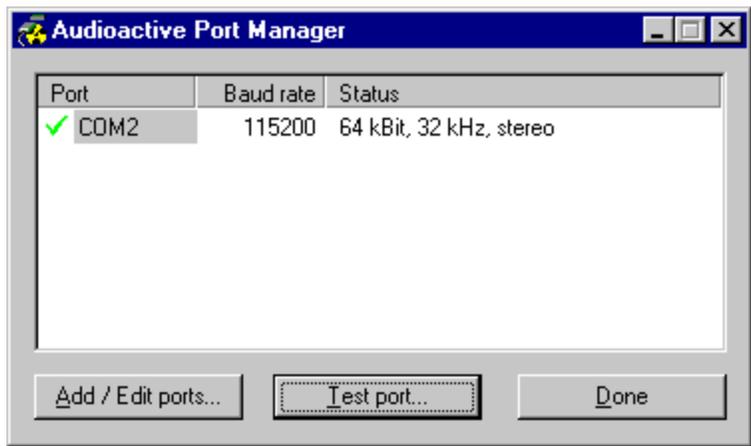
At this point, the setup is complete. Each encoder you added should now appear as an audio source for Windows Media Technologies' ASF Real-Time Encoder.

See Appendix A for more information on using the Windows Media Technologies' ASF Real-Time Encoder.

Audioactive Port Manager Application

This section applies when using the encoder with an RS-232 serial connection.

NOTE: Make sure you have all Audioactive Realtime Encoders connected to the PC and powered up while using the Audioactive Port Manager.

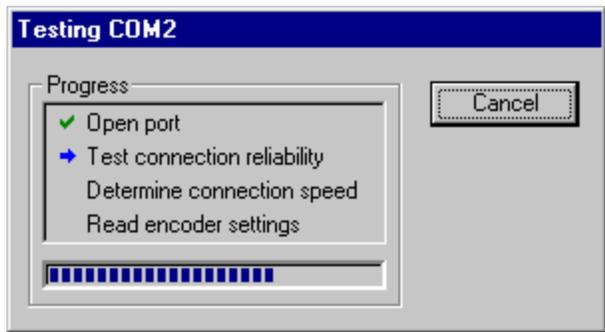


The Audioactive Realtime Encoder talks to the computer through an RS-232 communication port. The Audioactive Port Manager (AASETUP) application tests and configures the communication ports for you. AASETUP also modifies the BOOT.INI file to prevent Windows NT from incorrectly identifying the Audioactive Realtime Encoder device during boot-up.

AASETUP displays all communication ports registered with Windows. If the ports you intend to use are not in the list, click the Add/Edit button to add the ports.

Auto Setup

Select the first port where an Encoder is attached and click the Test Port button. AASETUP will test the port and report to you the results.



Four tests are performed on the port. The following table describes each test and some reasons why the test may fail.

Test	Test Purpose	Failure Reasons
Open Port	Test if port is available for use.	<ul style="list-style-type: none"> ◆ Port has not been setup in Windows yet. (Click on Add/Edit Ports... button to add port). ◆ Port is used by some other device (e.g. mouse).
Test connection reliability	Detect improperly connected cable or cables that are too long.	<ul style="list-style-type: none"> ◆ Cable may be too long or has a bad connection. Try using a shorter cable or tighten the connections.
Determine connection speed	Determine Audioactive Realtime Encoder baud rate.	<ul style="list-style-type: none"> ◆ The Audioactive Realtime Encoder is set to high speed (230,400 baud) but it is attached to a regular communication port that can use a

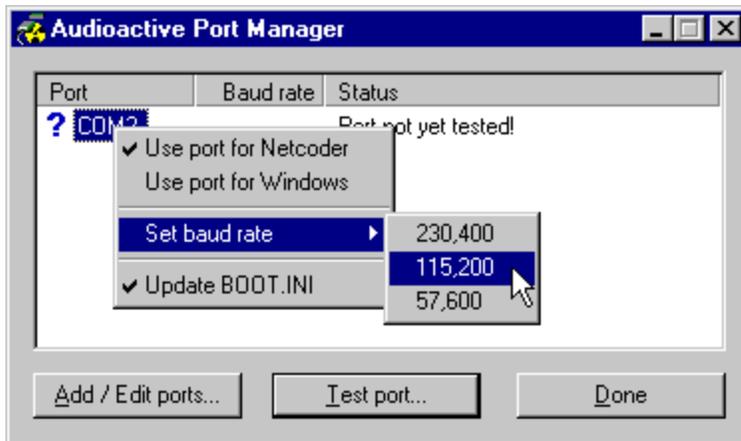
		maximum of 115,200 baud. Try setting the Audioactive Realtime Encoder baud rate to 115,200 baud or connect it to a high-speed serial card.
Read encoder settings	Read the Audioactive Realtime Encoder compression settings.	◆ Unlikely to occur. A device other than the Audioactive Realtime Encoder may be connected on this port.

If the port tests “OK” you should see the encoder settings (bitrate, sample rate and stereo/mono indicator) displayed on the screen. Repeat this process for each port where an Encoder is attached.

When all ports test “OK” the setup is complete. This encoder should now appear as an audio source for Windows Media Technologies ASF Real-Time Encoder.

Manual Setup

During port test AASETUP attempts to determine the baud rate of the encoder and connection reliability. If you would rather set the baud rate yourself, you should use manual setup. To use manual setup right click (or press the right arrow key) on the port name. A menu will pop up which allows you to set the port baud rate. The same menu allows you to reserve a port for Windows use.



Appendix A: Using Windows Media Technologies ASF Real-Time Encoder

Overview

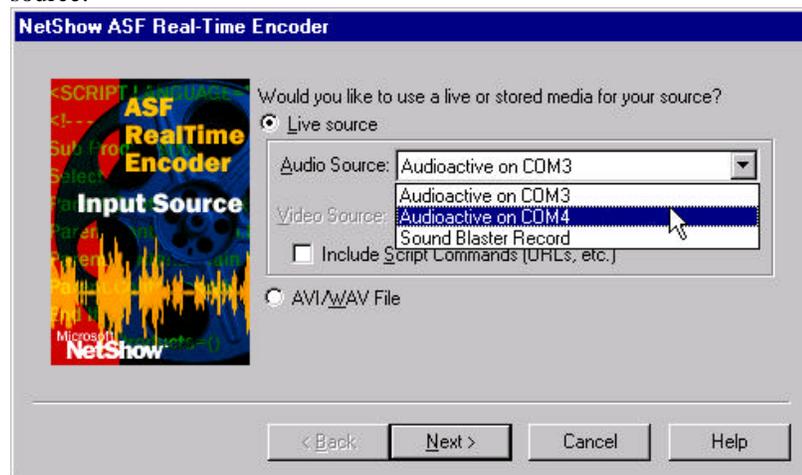
Microsoft Media Technologies uses the ASF Real-Time Encoder (RTE) to send live multimedia content to a Media Technologies' server. Using the Audioactive Multimedia driver, RTE can utilize the high quality compressed audio stream generated by the Telos MPEG Layer-3 Audioactive Realtime Encoder.

The following section describes how to use the Windows Media Technologies Real-Time Encoder with the Audioactive Multimedia Driver. For Windows Media Server help please consult the Microsoft online documentation.

ASF Real-Time Encoder

Audio Source

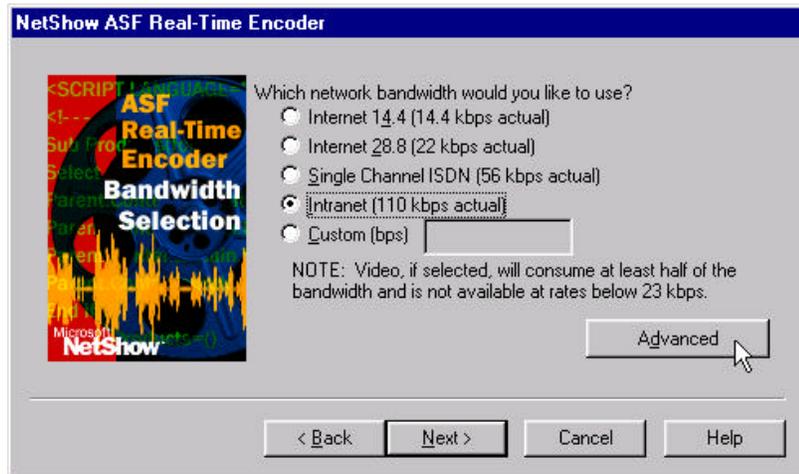
Start the ASF Real-Time Encoder and select New from the File menu. You will be presented with the following screen where you can select your audio source:



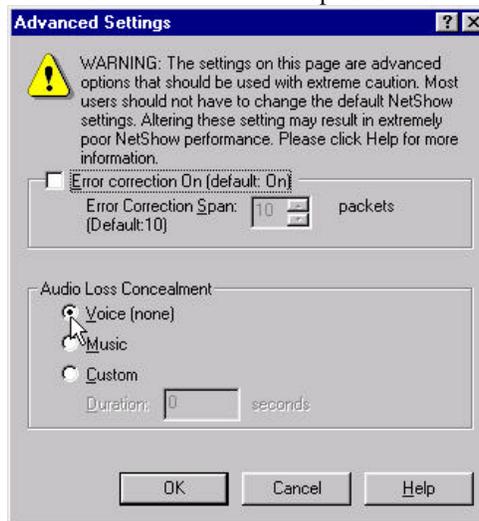
Select an Audioactive source from the drop down list box then click the Next> button.

Bandwidth and Error Correction

Select your available bandwidth from the following screen. The bandwidth selection made here limits the bitrate choices on the next screen.



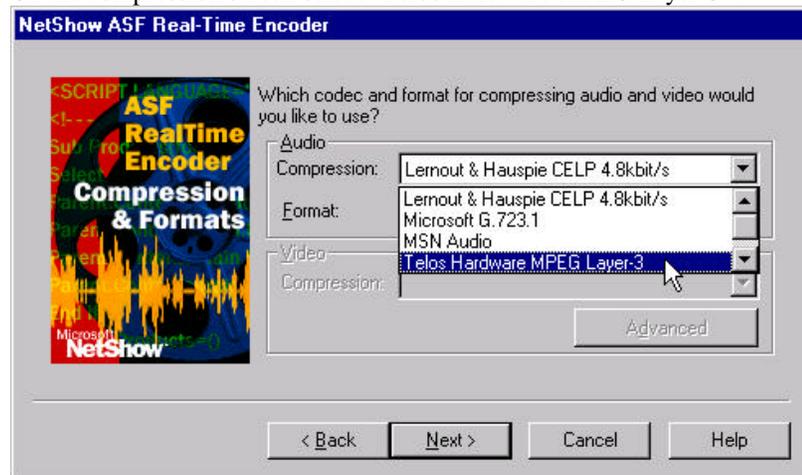
After selecting the bitrate, click on the Advanced button to set the error correction and concealment options. The following screen will appear:



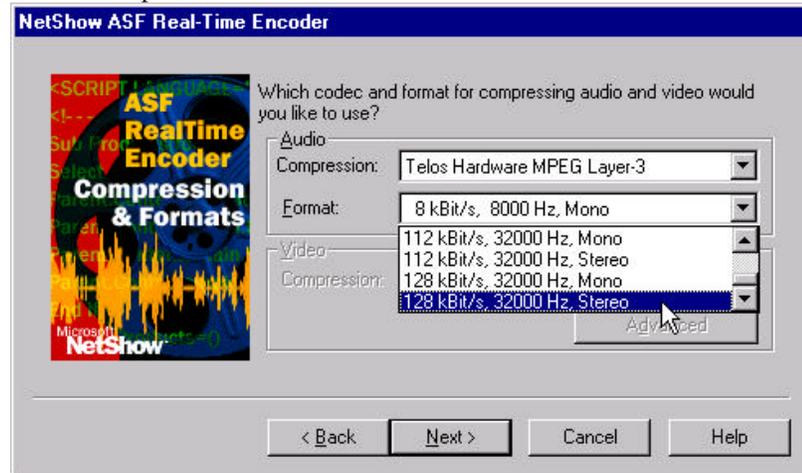
We recommend turning the Error correction off. The Audio Loss Concealment setting should always be set to None.

Compression

On the compression screen select “Telos Hardware MPEG Layer-3”.



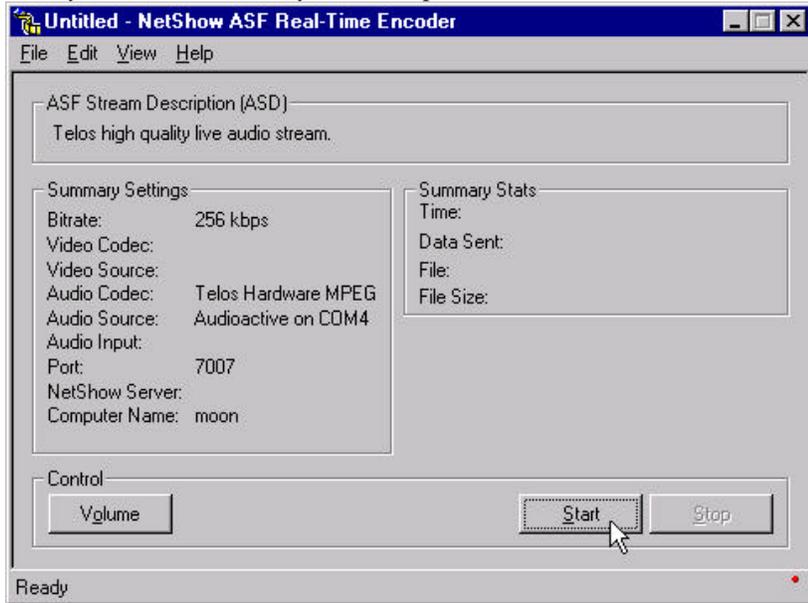
You will be presented with a list of available formats.



Select a format and make sure that the encoder display matches the format you selected. Use the encoder front panel control to adjust the encoder setting if required.

Start live source

After you click on “finish” you will be presented with this screen:



Make sure all your settings are correct then press start. You should see the “Data sent” counter increase after a few seconds. This indicates that the data is being sent to the Media Technologies’ Server.

If you use Audioactive Multimedia Driver on a port that does not have an encoder attached, you may get the following error:



If you get this error, check the following items, then try again.

- Encoder is turned on.
- The serial cable is plugged in securely.
- Encoder baud rate matches the serial port baud rate.
- The encoder is plugged in the port you selected.

Appendix B: Using MPEG Realtime Encoder as Stand-Alone Network Server

Unicast

The Audioactive Realtime Encoder supports a limited number of simultaneous connections in HTTP, TCP and all non-multicast Stream sender modes. The number of connections supported is dependent upon a given bitrate. The following numbers include both client connections and active unicast stream senders:

8 kbps - 24 kbps	15 clients (max)
32 kbps - 40 kbps	13
48 kbps	10
56 kbps	9
64 kbps	7
80 kbps	6
96 kbps	5
112 kbps	3
128 kbps	2

Notice: These connection limits are hardcoded, and attempts to connect clients or start new stream senders beyond these limits will fail.

Multicast

When enabled Multicast virtually allows an unlimited number of simultaneous listeners.

When a multicast IP address is specified in Stream Sender, the smallest TTL value appropriate for the application should be set to limit the potential of unnecessary “flooding” of distant routers. For testing, ect., a TTL of 1 (local subnet) is recommended.

Multicast IP addresses are in the range 224.0.0.1 through 239.225.225.225. Some address ranges are reserved for administrative applications and permanent assignments. Review the following list of reserved addresses prior to assigning an IP address: <http://www.isi.edu/in-notes/iana/assignments/multicast-addresses>

Care should be taken when choosing a multicast IP address for senders that have a TTL outside of the local subnet. Session tools such as SD should be used for scheduling use of addresses dynamically, and the IANA (<http://www.iana.org>) can actually assign permanent multicast IP addresses, although this is less common - usually permanent addresses are reserved for administratively scoped applications, etc.

To learn more about multicast addresses and how multicast routing works, check out: <http://hegel.itc.ukans.edu/topics/internet/internet-drafts/draft-i/draft-ietf-mboned-intro-multicast-03.txt>, <http://www.mbone.com> or www.ipmulticast.net

Appendix C: Serial Ports

Baud Rate and Fast Serial Cards

The Audioactive Realtime Encoder can communicate using either 115,200 baud or 230,400 baud rates. Using the 115,200 baud rate, the Encoder can output compressed audio up to 64 kbit/second. Higher bitrates can only be sent at 230,400 baud.

Most RS-232 serial cards commonly used in PCs support rates of up to 115,200 baud. If you wish to use bitrates higher than 64 kbit/second, you need to use a fast serial card. Fast serial cards support 230,400 baud rates or even higher.

We recommend the Turbo Serial I/O (TC-200) card from Byterunner (www.byterunner.com). The card does not require a special driver because it internally converts the baud rate from 230,400 to 115,200. This means that the card communicates with the encoder at 230,400 while the PC believes it is communicating at 115,200.

If you are using this card you should set the Encoder baud rate (though the Encoder front panel) to 230,400. The Audioactive Port Manager will set the port baud rate to 115,200 (which is converted by the card to 230,400).

Boot Time Hardware Detect

Windows NT scans for serial devices during boot-up. This scan is designed to detect a serial mouse plugged in any serial port. If Windows NT detects an Audioactive Realtime Encoder, it installs a mouse driver for that port. Windows NT will prevent any further access to the port not allowing applications to access the device.

To avoid this problem, the Audioactive Port Manager updates BOOT.INI file and tells Windows to skip the mouse scan on specified ports during boot-up by adding the /NoSerialMice parameter to the first line in the Operating Systems section.

Here is an example BOOT.INI file for a PC that uses COM2, COM3, and COM4 to communicate with Audioactive Realtime Encoders. Please note that your BOOT.INI file may look different.

```
[boot loader]
timeout=10
default=multi(0)disk(0)rdisk(0)partition(1)\WINNT
[operating systems]
multi(0)disk(0)rdisk(0)partition(1)\WINNT="Windows NT Server
Version 4.00" /noserialmice:com2,3,4
```

Appendix D: Omnia.net Audio Processor

Our sister company, Cutting Edge, announces the release of Omnia.net: the world's first digital signal processor for control of audio dynamics on the Web.

Pre-processing audio for the Internet? You bet.



Optimizing sound quality is as essential on the web as it is on traditional formats. Now *expert* audio processing is made possible through a fully-digital system that applies an advanced design philosophy yielding performance and flexibility previously unavailable to Internet audio.

Omnia.net is the result of years of experience in the professional broadcast industry, and research in the field of digital signal processing (DSP). When added to the signal chain before the hardware Audioactive Encoder, Omnia.net optimizes the bandwidth available on the Internet resulting in louder, cleaner, and even punchier sound.

The Omnia.net's processing resources are entirely configured by software and contained on a plug-in PC Card (PCMCIA). Even fundamental rearrangements of the system architecture may be accomplished by simply changing the card, or by remote download.

A second PC Card slot is available for a combined 10BaseT/modem connection for remote access. You can access Omnia.net through any Web browser. Using Netscape Navigator, Microsoft Explorer, or any browser of choice, the processor can be remotely controlled from any computer platform. Communication is via local serial RS-232, modem, 10BaseT, or even the Internet!

For more information on the Cutting Edge Omnia.net Audio Processor, go to www.nogrunge.com or call +1 216 241-3343.

Appendix E: Warranty

This Warranty covers “the Products,” which are defined as the various audio equipment, parts, software and accessories manufactured, sold and/or distributed by TLS Corp., d/b/a Telos Systems (hereinafter “Telos Systems”).

With the exception of software-only items, the Products are warranted to be free from defects in material and workmanship for a period of one year from the date of receipt by the end-user. Software-only items are warranted to be free from defects in material and workmanship for a period of 90 days from the date of receipt by the end-user.

This warranty is void if the Product is subject to Acts of God, including (without limitation) lightning; improper installation or misuse, including (without limitation) the failure to use telephone and power line surge protection devices; accident; neglect or damage.

EXCEPT FOR THE ABOVE-STATED WARRANTY, TELOS SYSTEMS MAKES NO WARRANTIES, EXPRESS OR IMPLIED (INCLUDING IMPLIED WARRANTIES OF MERCHANTABILITY AND FITNESS FOR A PARTICULAR PURPOSE).

In no event will Telos Systems, its employees, agents or authorized dealers be liable for incidental or consequential damages, or for loss, damage, or expense directly or indirectly arising from the use of any Product or the inability to use any Product either separately or in combination with other equipment or materials, or from any other cause.

In order to invoke this Warranty, notice of a warranty claim must be received by Telos Systems within the above-stated warranty period and warranty coverage must be authorized by Telos Systems. If Telos Systems authorizes the performance of warranty service, the defective Product must be delivered, shipping prepaid, to: Telos Systems, 2101 Superior Avenue, Cleveland, Ohio 44114.

Telos Systems at its option will either repair or replace the Product and such action shall be the full extent of Telos Systems’ obligation under this Warranty. After the Product is repaired or replaced, Telos Systems will return it to the party that sent the Product and Telos Systems will pay for the cost of shipping.

Telos Systems’ authorized dealers are not authorized to assume for Telos Systems any additional obligations or liabilities in connection with the dealers’ sale of the Products.