MaxxStream:

Delivering Quality Audio Content through the Internet and IP networks

WAVES

Table of Content

EXECUTIVE SUMMARY 3

MAXXSTREAM LEADS THE INDUSTRY BY:.. 3
MAXXSTREAM IS:.. 3
QUICK SURVEY OF MAXXSTREAM BENEFITS.. 4
ABOUT WAVES.. 5

WEBCASTING OVERVIEW 7

STREAMING AUDIO ENVIRONMENTS 7
STREAMING AUDIO ARCHITECTURAL STRUCTURE 7
STREAMING AUDIO OBSTACLES 9

MAXXSTREAM STREAMING SOLUTION 11

STREAMING ARCHITECTURE USING MAXXSTREAM 11
MEETING STREAMING REQUIREMENTS 12
MaxxStream and video streaming..15

MAXXSTREAM PROCESSING TOOLS 16

SYSTEM SPECIFICATIONS 21

MAXXSTREAM PCI / M100 / M200 21
MAXXSTREAM LX 22

Executive Summary

Waves' family of MaxxStream audio transmission processors truly represents a "next generation" technical achievement. Performing more than just processing, they are designed to provide a fully-integrated solution for the all-too-familiar problem of audio quality in Internet streaming.

The problem of audio quality has plagued Internet audio streaming since its birth. Spoken word can be unintelligible, music grainy and unpleasant, and conventional solutions are expensive to implement and maintain. MaxxStream makes Waves' world-recognized experience in developing high-end audio processors for the professional audio industry available to Internet streaming professionals.

MaxxStream is designed specifically for Internet streaming, of any type or sort. As a fully integrated audio capture, processing, and encoding workstation, it delivers superior full-spectrum sound in real-time, at any desired streaming bandwidth, and supporting any streaming format available today.

In order to provide a solution for all requirements, the MaxxStream family offers total scalability; from a single-stream radio station to a large-scale streaming installation; from live, on-site streaming to on-demand streaming; from production to encoding.

MaxxStream Leads the Industry by:

- Offering broadcasters and webcasters the highest possible audio quality content through any IP network - LAN, WAN, Intranet, or Internet
- Merging the whole process of audio capture, conditioning, and encoding into one unified workstation, providing easier control and management of operation.
- Offering the only non-proprietary solution that fits any production, broadcasting, and streaming environment, transparently supporting all popular digital media formats (RealMedia, Windows Media, QuickTime, MP3, etc.).
- Providing a future-proof flexible and scalable solution which can be quickly adapted to rapid technology changes.
- Allowing corporate media professionals to easily adapt content for quickly streaming information to employees, trainees, stockholders, or the press.

MaxxStream is:

- MaxxDSP DSP-based PCI audio card offered either as a stand-alone card or embedded in an integrated workstation.
- A feature-rich driver supporting streaming-specific requirements.
- MaxxStream software application that manages and controls the configuration and processing parameters of MaxxDSP.
- MaxxStream audio plugins/processors:
    - Q10 - 10 band paragraphic equalizer providing precision control of equalization.
    - L1 - Brickwall limiter and maximizer providing maximum level/resolution of audio signal plus peak limiting.
    - C1 - Compressor/Gate/Expander expert tool for wideband control of dynamics and automatic gain control.
    - S1 - A precise stereo imager providing control over joint-stereo levels.

- MaxxBass - A patented processor that utilizes proven psychoacoustic methods to deliver a significant bass enhancement through limited playback systems (PC speakers for example).
- DeEsser - An accurate de-essing tool allowing precision control and shaping of high frequency content or sibilants.
- AudioTrack - An all-in-one simple processor including a 4-band EQ, compressor, and noise-gate.
- C4 - A feature-rich multiband compressor/dynamics processor with mastering-grade degree of control.

Quick Survey of MaxxStream benefits

A Solution for All

Radio stations on-line service, or Internet-only stations, benefit by webcasting superior audio quality while lowering bandwidth costs. The MaxxStream integrated workstation is designed to turn incoming full-spectrum audio into a full-spectrum audio stream in real-time, at any target bandwidth.

Content archival, on-demand services, and encoding facilities gain by running pre-encoded content through MaxxStream before encoding or archiving.

Corporate communications, web conferencing services and integration providers, and distance learning services utilize MaxxStream to improve intelligibility in delivering the spoken word.

Lowering Streaming Costs

While MaxxStream’s superior audio quality is very easily demonstrated, the benefits of working with MaxxStream integrated workstation are not limited to broadcasting with crystal-clear sound.

MaxxStream drives streaming costs down by maintaining broadband-quality sound while paying for narrow bandwidth. Not only that; MaxxStream's support of multiple concurrent encoders and encoding formats allows one unit to be used to encode at multiple target bit-rates and formats, thus lowering costs per stream.

When producing content for streaming, MaxxStream saves production costs through unifying live audio capture and content-optimization into one real-time operation.

MaxxStream is an integrated solution that provides audio capture, processing and encoding in one rack-mounted computer, thus maximizing the use of precious machine-room space.

Future-Proof Solution

MaxxStream's flexible and scalable design and its non-proprietary support of encoding technologies allow it to provide MaxxStream users with a future-proof solution that can be easily modified as market or station requirements change. Any change translates to a simple one-step operation:

- Adding audio feeds translates to installing an additional MaxxDSP card to the system, not even requiring any software or architecture modification.

- Adding more output streams at any format or resolution translates to merely launching and running an additional encoder instance.
- Upgrading encoder technologies translates to a simple installation and running of the new encoder.

Quality Service

MaxxStream's superior audio quality allows broadcasters and webcasters to offer their client with a high quality service while keeping the cost of service low.

The Right Suite of Tools

MaxxStream provides the right suite of processing tools to fit any audio requirement, from solving encoder-specific audio artifacts to shaping a radio station's "signature".

Ease of Use

Working with MaxxStream is fast and easy. Creating a processing Rack for any application ranges from simple loading of one of MaxxStream's presets (more than 80 presets are included, organized by bandwidth and content-type requirements) to building a preset from scratch using MaxxStream's transparent hierarchic editing structure.

About Waves

Waves is the leading provider of DSP solutions for audio professionals in content creation and Maxx™ audio signal processing solutions for consumer electronics. Waves award-winning audio processor plug-ins are the technology and market share leader for thousands of audio professionals in content creation. These plug-ins utilize Waves proprietary DSP algorithms based on Waves psychoacoustic expertise.

Waves' Maxx technology dramatically enhances audio performance in consumer applications and has been licensed to several leading audio companies including Microsoft, Motorola, Samsung and Sanyo. Yamaha also purchased a minority position in Waves in 2000. Waves mission is to develop and provide solutions that enable unparalleled sonic quality for all audio applications.

Plug-In market leadership to the most quality sensitive customers

Waves has built a reputation for quality and technical leadership to the professional audio signal processing market. Waves is the market leader for software Plug-Ins used in the audio content creation process for music, movies and computer games with over 150,000 users worldwide. Waves offers both the broadest selection and the highest quality set of software solutions available. Its software solutions support over 20 different audio editing environments, both for native CPU operation (Windows and Mac operating systems) and DSP accelerator solutions.

Embedded Audio Systems

Waves offers hardware and software signal processing solutions in rack mount packaging for various applications providing more convenient use when a Digital Audio Workstation is not needed. The first of these products, the L2 Ultramaximizer includes high quality A/D and D/A converters and Waves renowned L2 limiter to increase the average signal level of typical audio signals without introducing audible effects.

MaxxStream Broadcasting Solutions for radio and Internet

Waves offers MaxxStream, the first streaming audio tool that integrates capture, processing and encoding into a single workstation. MaxxStream is complete hardware and software solution supporting Windows Media and Real Audio formats both reducing the cost of supporting multiple industry codec formats while dramatically improving the quality of the consumer audio experience
over the internet. The product also can be utilized as a powerful preprocessing tool to improve the quality of radio broadcasts.

Y56K DSP Processing Card for the Yamaha AW4416 and AW2816

Waves announced a partnership with Yamaha to develop the Y56K add in card for Yamaha's popular AW4416 Professional Audio workstation. The cards are loaded with six of Waves' proprietary audio processing functions.

Webcasting Overview

Streaming Audio Environments

Today's streaming arena contains many diverse applications and implementations of streaming technologies. The wide spectrum of streaming applications, ranging from radio stations streaming their broadcast content to corporate conferencing and training, plus the rapid evolvement of both market demands and supporting technologies, present any future or current webcaster with a confusing array of dubious possibilities.
A complaint frequently heard from webcasters is that designing a system requires the designer a thorough research and independent selection of appropriate technologies and that once the system has been designed it stands a high risk of becoming obsolete relatively quickly. This confusion leads in many cases to a partial and off-hand design of the streaming service, resulting in a very limited system that barely lives up to its requirements. In addition to the above it is noted that the above confusion leads in other cases to a decision to stay away from streaming until "we can figure out how the hell it works".

The mentioned obstacles have not prevented webcasters to go online with streaming services. The promise of delivering streaming audio - 24x7 service, crystal clear sound, revenue-generating interactive services, remote corporate training and conferencing - has attracted many companies and institutions of all sizes.
As an example, media companies are embracing the Internet as a medium of delivery to:

- Maximize program investments by re-broadcasting over the net
- Effectively capture audience demographic information
- Recapture audience members who are tuning out traditional media
- Expand audience membership around the globe

In addition, businesses outside the media and entertainment industries are embracing streaming media technology for a variety of applications such as:

- Employee Training
- Broadcasts of executive speeches
- Moving of closed circuit audio / video across data networks
- Remote product demonstrations
- Investor relations
- Customer support

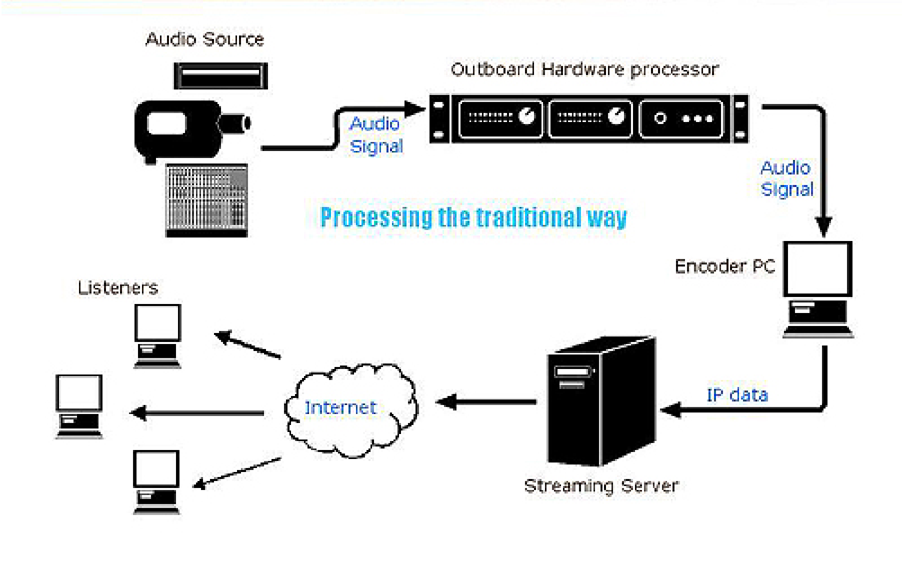
Streaming media across intranets or the Internet to user’s home and office PCs offers companies advantages when it comes to improving business processes and cutting communication costs. But there are also reasons (bottlenecks) why organizations are hesitant to stream content over the network, such as bandwidth and processing power limitations.

Streaming Audio Architectural Structure

Let us look at the typical live webcasting workflow process from the sound source to the listener's laptop or desktop computer. This paper will address the issue of audio service only. Issues of ad insertion, banner advertising, and other non-audio topics related to streaming are outside the scope of this paper. Additionally this paper will use the case of a radio online station as the reference model for this discussion.

The streaming process begins with a chain of rack mounted audio-dedicated processing hardware. This hardware provides the content/program producer the ability to manipulate and prepare the media to become digitized.

Once the content has been prepared it is converted from the analog domain to digital and routed through a streaming media encoding application on a CPU intensive machine in preparation for distribution across the network. The content is then moved across a hub (such as ISDN) or modem that connects to a more powerful computer where the streaming media server software runs. The server is connected via high-bandwidth connection to the Internet where content providers can webcast digital content across the network to their audience.



The analog or digital audio signal is conditioned by ad outboard processor, captured by a sound card, passing the content to the encoder application after being digitized (which taxes the machine's CPU.) Once the audio is encoded by the encoder PC, it is delivered to the streaming media server through the network (or modem) and then made available for reception by the end user's computer.

A simpler architecture of the above is realized in low-cost operations where the audio signals are not processed but are directly captured and digitized by a low-quality sound card installed on a PC that also performs the encoding.

It is interesting to note, that in the current situation the server acts as the middle tier in the process, replicating the stream coming from the encoder for each listener. This explains why the encoder to server connection can be of lower bandwidth than from the server to the Internet connection.
In this entire process, available network bandwidth, encoder PC processing power, and consistency of sound during the digitization process become major concerns that force organizations to hesitant regard of the streaming of media over the networks.

Streaming Audio Obstacles

Listed below are several obstacles that require evaluation by content producers looking at the option of streaming media content across IP networks and Internet.

Cost of Bandwidth

Bandwidth does not come cheap and its cost needs to be configured into the monthly streaming budget. Although the cost of bandwidth between the encoder workstation and streaming server is relatively low as this connection is usually low bandwidth, the cost of bandwidth from the server to the Internet (being wide-band) is expensive.

This cost pushes webcasters away from streaming at wider bandwidths resulting in low-quality audio service and a limited number of streams, thus serving only a narrow spectrum of the potential audience.

Streaming Content to a Wide Variety of Recipients

Bandwidth costs and the cost of the hardware required force content producers to limit the number of streams they cast into the Internet. From the content producer point of view webcasting to a wide variety of potential listeners means streaming at various formats (RealMedia, Windows Media, etc.) and resolutions (56k modems, ISDN, DSL, etc.). This dictates a lot of computers, processing gear, and a big headache to install, maintain, and update.

The result is again a limited number of streams, usually at low quality.

Sound Quality

The poor sound quality of Internet webcasts has been discussed before. This bottleneck can be overcome by streaming at higher resolutions (96kbps instead of 32kbps for example) or by conditioning the audio content prior to its encoding.
Unfortunately, these solutions don't come cheap. Higher streaming resolutions involve a higher cost of bandwidth, and processing gear is expensive and does not offer enough flexibility for future requirements adaptation.

An additional issue related to sound quality is the type of processing applied to the audio signal prior to encoding. There are several cases where a broadcaster, already transmitting content on air, has used his on-air FM pre-processors to feed the streaming encoder. The result, although better than a non-processed signal, is not much better, since the processing applied to FM signals is not optimal for streaming. The reason is that streaming encoders utilize a totally different model for encoding than FM modulation. Most streaming encoders use various derivatives and implementations of an MPEG codec, based on psychoacoustic masking knowledge-base. These codecs behave differently than RF modulation and therefore present different artifacts to the audio signal.

Hardware Limitations

Designing an encoding and streaming station requires taking the involved hardware limitations in mind.

Digitizing and encoding an audio stream (let alone video) taxes the encoding computer's CPU. Since encoding is CPU intensive, providing several streams requires several encoding computers. Adding processing to the computer's CPU requires very fast computers, and probably even several of them.

Since audio must be captured and digitized by the encoding computer, a sound card must be installed. The majority of sound cards can only deliver one instance of their incoming signal to the system. If the content producer tries to connect the same device to a second application (another encoding application for example) the system alerts that the device is being used by another application. This limits the number of output streams to the number of sound cards installed - one sound card per one stream.

Installation and Maintenance

Combining the above restrictions quickly becomes a third problem - machine-room space and management.
Working the "traditional" way means using several audio hardware processors, multiple stacked computers, multiple sound cards, miles of cables, and additional supporting audio patch-bays or routers - all with their own interfaces and connectivity requirements.
Maintaining this type of setup means either several control systems utilizing different interfaces (traditionally audio processing gear has front panel interface, encoding computers use monitors, keyboards, and mouse, etc.) or sending a technician to the machine-room, hoping he'll come back.

The above setup carries difficult monitoring and maintenance procedures, complicated installations of new equipment, let alone updating the setup, all translating to technician time and cost.

Future Proofing

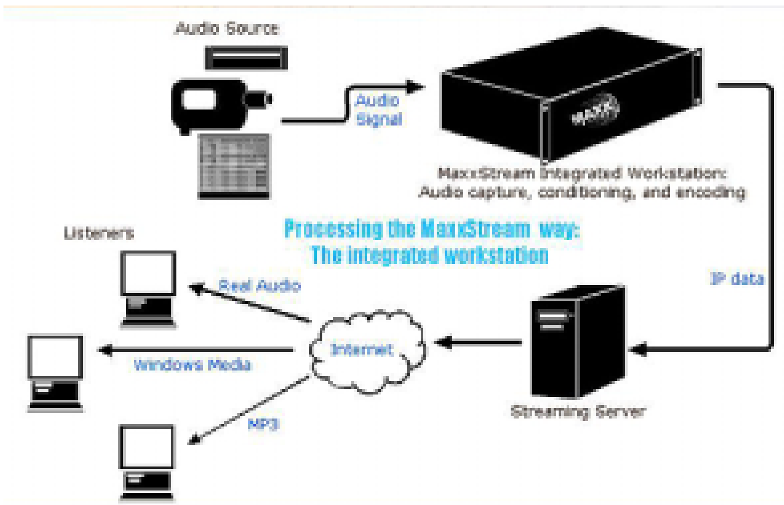
Streaming technologies evolve quickly. New versions of encoders and new codecs appear at the market at a relatively quick pace. This reality presents any content producer with the alarming prospect of his system becoming obsolete or losing a competitive edge only after a short while.

A content producer wish is to have a system that is as future proof as possible given the liquid state of streaming technologies. It would be optimal from the system aspect to design a system that could be easily updated and/or adapted to accommodate changes in both technology (new encoder versions for example) or market demands (for example, serving a new target audience requiring other resolution streams).

MaxxStream Streaming Solution

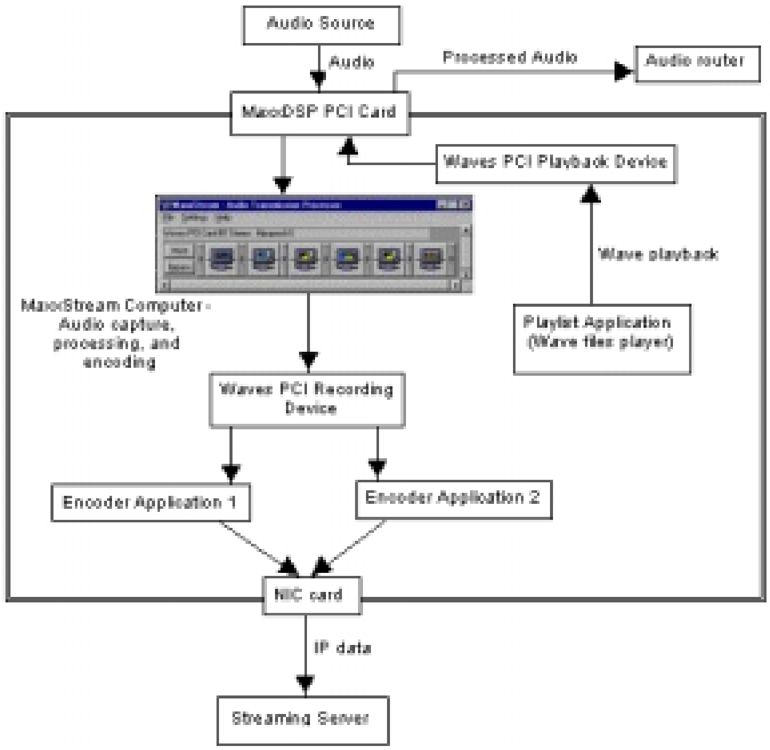
Streaming Architecture Using MaxxStream

MaxxStream is the solution for easy distribution of live broadcast content to the Internet or a company's Intranet. It provides the means to capture, process and encode many digital streams on a single PC running Windows NT/2000/XP. It's possible to transmit up to 8 separate mono audio signals, or 4 stereo on the same PC. The solution is powered by Waves Digital Audio Processors which allows users to deliver the best sonic quality possible with today leading technologies while remaining open and flexible to tomorrow's digital solutions of your choice.



MaxxStream significantly improves the entire webcasting process by providing an integrated workstation whose components are optimally utilized (DSP-based processing of audio signals, CPU-based encoding). Once the content is digitized, MaxxStream's DSP card instantly routes content to several streaming media encoders at one time. MaxxStream provides a full suite of powerful plug-ins allowing content providers to optimize the digital content for broadcast across the Internet so users can enjoy their listening experience.

The dedicated PCI audio card equipped with a Motorola DSP coprocessor gives MaxxStream the power to process and optimize the transmission signal, without taking resources from the main CPU, leaving it free to deal with the encoding process. The software processors replace the need to invest in large quantities of expensive outboard gear.



Audio signal is captured by the MaxxDSP PCI card and processed by MaxxStream's processing chain. Then the processed audio is published to MaxxStream's driver (Waves PCI Recording Device) where it's picked up by any encoding application, encoded, and sent through the computer's NIC card as IP data to the streaming server. The processed audio is also simultaneously sent through MaxxDSP card's analog and digital outputs. As the above shows, MaxxStream is also able to receive input from any wave playback application running on the same computer such as radio automation software.

MaxxStream's signal allows users to feed digitized content to multiple encoders either in file or live streams format from a Windows recording device. The user can replicate the audio signal allowing connecting more than one application to the same stream. This way it is possible encode the same signal to more than one destination stream, for example - one transmission for DSL modem users and another for 56k modem users.

Meeting Streaming Requirements

MaxxStream has been designed with streaming in mind. In this section we'll discuss various streaming requirements or obstacles and the way they are addressed and solved by MaxxStream.

Bandwidth Considerations

As explained above, bandwidth is directly related to cost on one side and the quality of audio service on the other. The only reason of streaming at a wider bandwidth is to provide a higher quality of audio service.

MaxxStream addresses this obstacle by applying signal processing on the audio before it is encoded. Its processing presents the encoder with the best quality of audio signal it requires to produce the best quality audio stream. The quality difference can be viewed in two ways. One way to look at it is to realize that the quality of audio, when processed, is wideband-grade using the same bandwidth as before, thus obviating the need to stream at a higher resolution. Another way of looking at this equation is to consider that, with processing, it is possible to maintain the same quality of audio service while streaming at a lower bandwidth than is currently required.

Streaming Content to a Wide Variety of Recipients

It is an understood wish of the content producer to be able to stream his content to the widest possible variety of potential listeners. To realize this wish the encoding and streaming system should be able to stream multiple formats (RealMedia, Windows Media, QuickTime, MP3, etc.) and multiple resolutions (20kbps, 32kbps, 64kbps, 96kbps, etc.), all in real time and concurrently. It might be the case that at a certain point in time the system is designed to stream only a limited set of streams but be specified to openly provide any future stream-type that will be required of it.

MaxxStream addresses this requirement in two ways. First, MaxxStream integrates the audio capture, processing, and encoding into one PC machine, be it a factory-built and pre-configured M system (MaxxStream M100 or M200 system) or an existing encoding PC with MaxxStream PCI (or MaxxStream LX) audio cards. As an integrated system it opens the possibility to use its specially designed driver for supporting multiple concurrent encoder instances.
Secondly, MaxxStream's driver is specially designed to allow an unlimited number of "client" applications to connect to its driver and receive the processed audio signal. The driver provides each encoder instance with the processed audio signal sample-rate converted to the specific sample rate requested by the specific encoder instance (the sample rate request of the encoder is derived from the streaming resolution profile the encoder is set for). All of this occurs in the background, no user intervention is required. The result is that one MaxxStream system can encode one audio feed to an unlimited number of streams, independently of format or resolution/bit-rate.

Sound Quality

Any content producer would like his stream to sound as good as possible in given bandwidth restrictions.

The sound quality issue is addressed by MaxxStream by applying the right type of processing to the audio signal before it is encoded by the encoding application. We will elaborate on this issue further on but will only mention here that the processing tools MaxxStream contains address the specific audio artifacts produced by the severe data reduction applied by the encoder.

Additionally, sound quality is raised by using high quality converters on MaxxDSP sound cards and by utilizing Waves world-renowned digital audio processors for processing.

Hardware Limitations

The concept of the integrated capture and encoding workstation, realized in MaxxStream is applied here to address and solve streaming hardware limitations.

MaxxStream is a DSP-based audio processing application. All the processing of audio signals occur on the DSP chip onboard the MaxxDSP card. The result is that all of the computer's CPU is left alone to handle the CPU intensive encoding tasks, making an efficient use of CPU resources. For reference, MaxxStream takes about 0.03% of CPU resources on a standard PIII 500mHz. This figure is consistent and independent of the processing type.

Combining this aspect of MaxxStream with MaxxStream's ability to simultaneously handle multiple encoder instances results in the optimal use of any available CPU. For example, one M100 system, occupying 1UR rack space can output a minimum of 6 streams (variably depending on output resolutions) making maximal use of both the CPU hardware (including involved sound cards) and rack space.

Installation and Maintenance

The issue of installation and maintenance is again solved by MaxxStream's integrated "nature". A MaxxStream workstation presents its user with one standard Microsoft Windows-based interface that allows control and administration (either local or remote) of all audio processing aspects and encoding parameters. All administration and access of all aspects of the system use standard Windows technologies. Another aspect of this unified interface is the ease of maintenance and updating - MaxxStream appears as just another standard Windows computer on the network.

In addition, MaxxStream presets are small Windows text files, allowing a webcaster to tune one processor and then distribute this present either by email or network to all MaxxStream units present.

Real time Operation

A streaming system should be as real time as possible. The traditional process of recording material, sending it off-line to processing and encoding is less desired.

MaxxStream's capture, processing and encoding is totally real time allowing it to be used even as an encoding station in the field for real time webcasts, or for real time IP-based conferencing.

Future Proofing

The issue of future proofing the MaxxStream workstation against the changing tides of the streaming arena is solved by making the system as flexible as possible, meaning that new requirements are addressed either by software installation or by a simple change of one component:

- More streams - If the system is required to publish more output streams, then the user has only to launch an additional encoder instance and start streaming. No hardware or software changes/modifications are required. In extreme cases where launching another encoder will run the risk of overloading the CPU, only that component will require upgrading.

- More audio feeds - If for example a MaxxStream system is required to serve an additional client, meaning it needs to receive more audio feeds as input, then it's only an issue of inserting another MaxxDSP card to the system. No software or other hardware modifications are required, not even driver re-installation (the driver recognized the number of installed cards at system boot time).
- New encoding technologies - Since MaxxStream performs its "magic" prior to the encoder and is encoder-agnostic, implementing a new encoding technology requires just the installation of the encoder software on the MaxxStream system.

Ease of Use

Ease of use is mentioned here as a part of the requirement set for streaming. One reason is that a webcaster has to sometimes overcome a stiff learning curve before successfully streaming. In that aspect any system that is simple and intuitive reduces this learning curve, thus saving time and effort. Another reason is that a machine that is difficult to operate is a machine that is either left alone or used only to a limited degree.

MaxxStream has been designed for simplicity of operation. It includes more than 80 presets, organized by target bandwidth and content-type, allowing a webcaster to load a preset, start streaming, and forget about it. In addition MaxxStream supports a simple and intuitive user interface making creation or editing of presets a snap.

MaxxStream and video streaming

When regarding MaxxStream's application for video streaming the same bandwidth vs. quality equation, discussed above, comes into play. When encoding an audio/video signal, the amount of bandwidth dedicated for each of signal's portions (audio and video) determines their respective quality. It is desired that both signals have the best quality at any given overall bandwidth. That given, MaxxStream allows the video encoder to use more bandwidth while keeping the high quality of the audio portion of the signal.

MaxxStream Processing Tools

Waves processors included in MaxxStream are:

Q10 Paragraphic Equalizer

The Q10 provides precision control of equalization from subtle adjustments to extreme complex filters.
Q10 provides from 1 to 10 bands of equalization, configurable as either two independent mono equalizers or as a precisely ganged stereo equalizer. Each of the 10 bands can be true parametric, high/low shelves, or high/low pass (cut) filters, and to ensure optimum fidelity, each is noise-shaped for the best possible signal-to-noise ratio.
Q10 includes equalization presets including de- and pre-emphasis curves suitable for AM, FM, and J17 transmissions.

In the context of streaming conditioning, Q10 is generally used for frequency limiting to prevent aliasing artifacts and to compensate perceived brightness for high frequency loss at the encoding stage.

C1 Parametric Compander

The Waves C1-Compressor/Gate is a studio-quality stereo dynamics processor and a "parametric compander". The C1 is also a fully featured dynamic filtering processor capable of many unique effects including sound sweetening, enhancement applications, and applications requiring correction of sound quality problems. For maximum flexibility and efficiency, the C1 breaks into smaller components, such as the C1comp and C1gate, or the C1c/sc (comp with sidechain).

C1 is used by MaxxStream as both a noise-gate (C1gate) to filter out unwanted ambient noise and as an AGC (Automatic Gain Control) with slow attack and release times and long lookahead.

L1 Ultramaximizer

The L1-Ultramaximizer is a highly sophisticated audio processing tool combining an advanced peak limiter, a level maximizer and a high performance re-quantizer based on the implementation of Michael Gerzon's IDR (Increased Digital Resolution) noise-shaped re-dithering process.

L1's lookahead peak limiter provides the broadcasting/webcasting engineer with the capability to increase the audio signal's resolution and production levels with precise control (and selectable dithering options).

While the operation of conventional limiters is well understood, the limiter section of the L1-Ultramaximizer is capable of a very fast, overshoot-free response, and once the limiter threshold has been set, the user can then go on to define the actual peak level that the processed signal will reach. Once set, limiting and level rescaling becomes a one-shot process. Using the L1-Ultramaximizer, it is generally possible to significantly increase the average signal level of a typical audio signal without introducing any audible side effects.

L1 is used by MaxxStream to perform two functions. First, as a maximizer it is used to raise the signal level to both generate a loud signal and to supply an audio signal with the highest signal-to-noise ratio to the encoder. Second, as a brickwall limiter, it is used to protect the encoder from overshoots.

S1 Stereo Imager

The S1-Stereo Imager combines a number of classical stereo processing techniques with Waves' intuitive user interface to provide a powerful and unique 'stereo toolkit'.
S1 is based on recognized engineering principles and does not purport to create a three-dimensional soundfield from a two-loudspeaker system or to create a pseudo stereo master from a mono source. Because S1 is based on established audio engineering principles, it offers a high level of mono compatibility with minimal side effects.
S1 provides the means to readjust the stereo level-balance of any audio signal and has the ability to dramatically widen an existing stereo image, again without introducing significant side effects.

For streaming purposes, the S1 provides a way to control the quality of the encoded stream. A streaming encoder (or actually any MPEG-based encoder) looks at the difference in content between the left and right audio channels. The higher the "similarity" between the two channels, the more efficient the encoding process is, resulting in a higher-quality signal. The S1 allows the user to control the "amount" of stereo in the signal (or amount of joint-stereo) raising the level of efficiency in the encoder.

S1 is a MaxxStream-unique processor, not found in other broadcast processors.

C4 Multiband Parametric Processor

You may think of the C4 as a multiband compressor with parametric adjustments, or as a 4-band dynamic equalizer. Both concepts are true, but the interface looks more like an equalizer with a moving line!

Waves' unique DynamicLine™ display shows the actual gain change as an EQ display. We have taken the gain reduction metering function and merged it with the crossover display for a very intuitive interface that gives you a precise feedback of what is going on.
C4 can be considered as a dynamic equalizer capable of gentle compression, expansion, limiting, and EQ, independently and simultaneously,

The C4 has a phase-compensated crossover with a flat frequency response when set to nominal values.
Such a crossover design is important to avoid any undesirable coloration to the sound and to eliminate artifacts and pitch-shifting effects between bands as their gains move independently. Both crossover points and their Q value are not fixed but user controlled.

For conditioning purposes, C4 allows a high level of frequency-related dynamic control.

De-Esser

DeEsser is an audio plug-in for attenuating higher pitched frequencies such as those found in 'ess', 'shh' and 'chh' sounds. DeEsser features very sharp filters in the sidechain and the choice of Wideband or Split audio paths.

DeEsser is used for both processing speech content and to attenuate the audibility of unpleasant high frequency artifacts created by some of the streaming codecs.

DeEsser is a MaxxStream-unique processor, not found in other broadcast processors.

MaxxBass

MaxxBass is patented psycho-acoustic bass frequency extension technology (U. S. Patent #5,930,373) that provides perceived bass frequency response below the physical speaker cutoff in an audio system. MaxxBass utilizes the principle of the missing fundamental, which creates the sensation of low frequencies by generating a carefully calculated series of harmonics designed to simulate the auditory experience caused by the missing fundamental pitch. These harmonics extend the virtual frequency response up to $1^{1}/_{2}$ octaves or two-thirds below the physical speaker cutoff frequency without perceived distortion or increased power consumption.

The MaxxBass processor is used in a MaxxStream processing chain to compensate for low frequency loss and maintain deep bass content in any listening environment, which in the case of streaming is usually at home or at the office, using small and cheap PC speakers.

MaxxBass is a MaxxStream-unique processor, not found in standard broadcast processors.

AudioTrack

AudioTrack contains a fully parametric 4-band EQ, a very versatile compressor, plus a noise-gate, with quality and control.

AudioTrack provides these three tools in one simple screen, making it very straightforward and easy to use. Its controls and monitors are intuitive, giving you an overall picture at a glance.
AudioTrack algorithms are optimized for computational efficiency.

System Specifications

MaxxStream PCI / M100 / M200

Audio Inputs
Analog:
Balanced - 2 XLR connectors (breakout). +22dBu – professional
Unbalanced – 2 RCA connectors (breakout). +8.9dBu – consumer
Digital:
AES/EBU - XLR connector (breakout), supports up to 24-bit @ 96kHz.
SPDIF – RCA connector (breakout), supports up to 24-bit @ 96kHz.
Input Impedance: 10kΩ

Audio Outputs
Analog:
Balanced - 2 XLR connectors (breakout). +22dBu – professional
Unbalanced – 2 RCA connectors (breakout). +8.9dBu – consumer
Digital:
AES/EBU - XLR connector (breakout), supports up to 24-bit @ 96kHz.
SPDIF – RCA connector (breakout), supports up to 24-bit @ 96kHz.
Output line drive: 50Ω

Analog-to-Digital Section
Resolution: 20 bit
Sample Rates: 44.1 / 48 / 88.2 / 96 kHz
Frequency Response: 20Hz-20kHz ±0.1dB
THD+Noise: 0.05% @ 1 kHz

Digital-to-Analog Section
Resolution: 20 bit
Sample Rates: 44.1 / 48 / 88.2 / 96 kHz
Frequency Response: 20Hz-20kHz±0.07dB
THD+Noise: 0.055% @ 1kHz

DSP Section
Resolution: 24 bit
Motorola 56301 ONYX 400MIPS

Factory Presets

Bandwidth Presets
Low – connections below a 28.8 kbps analog modem
Medium – more common 28.8 - 56 kbps contemporary connections
High - highest connections using ISDN or faster connection types.

Content Type Presets
General – Signals of mixed types, voice & music of various genres.
Pop/Rock - popular contemporary music styles
Classical - Classical music signals
Speech – For enhancing intelligibility of news and event broadcasts

Stereo and Mono Destinations
This is only for stereo rack presets, as transmitting a stereo source in mono will improve its quality to the listener by 50%, while transmitting mono signals in stereo would probably degrade the quality for the listener.

Software Processors

AudioTrack, L1 Ultramaximizer, Q10 Paragraphic Equalizer, C1 Parametric Compander, S1 Stereo Imager, C4 Multiband Parametric Processor, De-Esser, MaxxBass.

MaxxStream LX

Audio Inputs
Stereo analog - 2 RCA connectors, -10 dB consumer line level adjusted.
Digital SPDIF - RCA connector, supporting up to 24-bit @ 96 kHz (SPDIF standard levels).
Input Impedance - 10 kΩ

Audio Outputs
Stereo analog - 2 RCA connectors, -10 dB consumer line level adjusted.
Digital SPDIF - RCA connector, supporting up to 24-bit @ 96 kHz (SPDIF standard levels).
Output line drive - 1kΩ

AD section
ADC resolution - 20-bit
ADC Sample rates - 44.1 kHz, 48 kHz, 88.2 kHz, 96 kHz.
Frequency response - 20 Hz - 20 kHz ±0.05 dB
THD+Noise - 0.05% @ 1 kHz

DA section
DAC resolution - 20-bit
DAC Sample rates - 44.1 kHz, 48 kHz, 88.2 kHz, 96 kHz.
Frequency response - 20 Hz - 20 kHz ±0.1 dB
THD+Noise - 0.07% @ 1 kHz

DSP: Motorola 56301 ONYX - 80 MIPS

Factory Presets

Bandwidth Presets
Low – connections below a 28.8 kbps analog modem
Medium – more common 28.8 - 56 kbps contemporary connections
High - highest connections using ISDN or faster connection types.

Content Type Presets
General – Signals of mixed types, voice & music of various genres.
Pop/Rock - popular contemporary music styles
Classical - Classical music signals
Speech – For enhancing intelligibility of news and event broadcasts

Stereo and Mono Destinations
This is only for stereo rack presets, as transmitting a stereo source in mono will improve its quality to the listener by 50%, while transmitting mono signals in stereo would probably degrade the quality for the listener.

Software Processors

AudioTrack, L1 Ultramaximizer

Waves Contact Information

North America:

Waves, Inc., 306 West Depot Ave., Suite 100, Knoxville, TN, 37917,

Tel: (865) 546-6115

Fax: (865) 546-8445

MaxxStreaminfo@waves.com

www.maxxstream.com

www.waves.com.

Israel:

Waves, Ltd., Azrieli Center 1, 21st floor, Tel-Aviv, Israel, 67021

Tel: 972-3-6081648

Fax: 972-3-6081656
MaxxStreaminfo@waves.com

www.maxxstream.com

www.waves.com

MaxxStream International Sales Management

Mick Olesh: +972-3-6081754

####