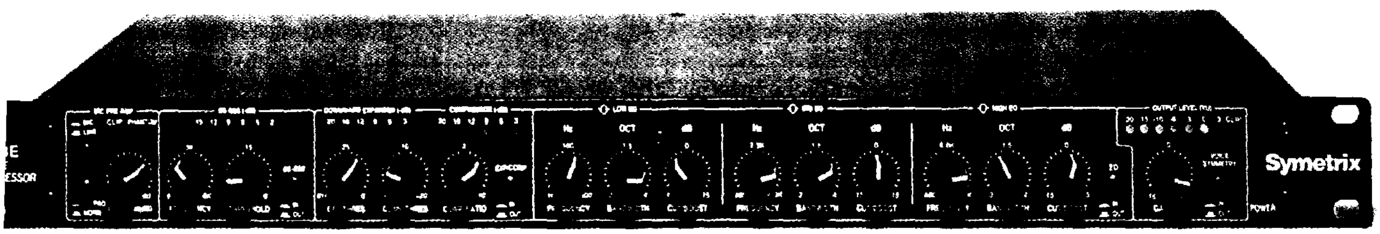
Preprocessing Streaming Audio



FRANK BEACHAM

Streaming media — especially audio — is revolutionizing the Internet. Technologies such as lNetwork's RealSystem, Microsoft's Show and Apple's QuickTime now let ually anyone operate a global media out- or less than the cost of a weeklong vaca- i in New York City. Because bandwidth itations still hamper video over the rnet, audio remains the sweet spot in aming media.

d, because most our computers don't reproduce h-fidelity sound, ce-based programming continues e the most successful and satisfy- form of streaming media.

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Yet just because the delivery quality of audio has dramatically improved in the t year, it does not necessarily mean the ces coming from your personal computer sonically pure. Too many webmasters hazardly encode any sound sent their without consideration to optimizing it Internet delivery.

RealNetworks, whose RealAudio tech- ogy dominates with more than 85 percent nternet users, offers an excellent on-line rial on how sound producers can maxi- e the quality of streaming audio. Obviously, all the basic rules apply. Good microphone technique is critically important, as is correctly setting levels and using high-quality recording equipment. But what's often overlooked in producing streaming audio is preprocessing.

The recommended preprocessing of voice files for streaming media falls under the two main categories of gain control, which include noise gating, compression and limiting, and equalization. All these functions can be accomplished through audio editing applications and plug-ins on the computer desktop.

In my experience, however, some software-based preprocessing tools can be quite limiting and cumbersome. Some can play only short loops of your program through the computer for evaluation and tweaking. Because sound applications are generally demanding of your computer's processor, they can literally take over and prevent you from doing other jobs with your PC.

It seemed to me — at least for this multi-staged task — a mix of hardware components and software might be in order. Thus, my attention turned to voice processors. Today's voice processors usually combine four or more functions in a single box. Though not designed for preprocessing of streaming audio files, it seems these devices had the right combination of features and cost to fit the job.

We worked with the Symetrix Model 528E, an analog voice processor priced at $749, for the preprocessing of RealAudio files. The 528E combines a microphone preamp, de-esser, compressor/limiter, downward expander and parametric EQ. We provided the 528E a signal from a MiniDisc player (bypassing the microphone preamp) and fed the unbalanced output to the audio input jack on an Apple Powerbook G3 computer. We used Macromedia's SoundEdit 16 application for recording to hard disk. The balanced output of the 528E fed a headphone amp for critical preview monitoring of the voice processor's setup.

The Symetrix box did the job very well and gave us a level of convenience and control beyond most of the software-based preprocessing solutions we've previously tried. Symetrix's superb installation manual guided us through the device and gave us the starting points for each function. We simply tweaked the controls until each program sounded best and then recorded the audio to the computer's hard drive. It was that easy.

The first stop on the 528E is the de-esser. It selectively removes the high frequencies from the input signal when sibilant sounds are present and exceed the threshold level. It's not an essential preprocessing function for streaming media but can be useful when you need it.

Next is downward expansion, a function that can lessen background noise that becomes audible during pauses in the audio. We simply turned the threshold control so the gating or expansion occurs when there is no desired audio; but not so high that the beginning of words we want to hear are chopped off.

As for compression, we optimized for RealAudio's medium bit-rate codecs for 28.8 modems, the most common configuration on home computers. A ratio of 2:1 and 4:1 is generally about right for speech. With low bit-rate codecs for 14.4 modems, compress 4:1 to 10:1. For high bit-rate codecs (ISDN), use very mild compression, if any.

EQ settings also depend on the codec. Since RealAudio's low bitrate codecs discard much of the high end, it's important to boost the midrange frequencies in the 2.5 kHz area to make speech more intelligible. For medium bit-rate codecs, we found a modest boost in the mid frequencies worked best. For high bitrate codecs, properly mastered programs should need no additional EQ. There are no set rules, however. It's best done by ear.

The only remaining preprocessing we did before encoding to RealAudio was to "normalize" the sound file. This is a function in which the computer's processor determines how much it can increase the loudness of the file without causing distortion. We did this on the Powerbook, since the voice processor has no normalize capability. Normalization (to 95 percent in the case of RealAudio) is the last step of the process prior to encoding.

We are now convinced that a combination of hardware- and software-based preprocessing tools simplifies the process of producing streaming audio. A voice processor, like the excellent Symetrix 528E, warrants serious consideration.

To read RealNetwork's "Getting the Best Sound Out of Real Audio," visit: *www.real.com/devzone/library/audiohints.html*. Symetrix's Web site is at *www.symetrixaudio.com*.

Frank Beacham is a New York City-based writer and producer. Visit his Web site at: www.beacham.com. *Mail:* 163 Amsterdam Ave. #361, New York, NY 10023. *E-mail:* frank@beacham.com.