

18.0 Web Broadcasting Applications

18.1 Why process the Audio?

Listenability

- Consistent Level
- Consistent Tonal Balance
- Less Distortion
- Improved Intelligibility
- Less Listener Fatigue
- Longer Listener Retention

Designed Sound

- Adjustable Peak and Average Program Density
- Adjustable Dynamic Equalization
- Adaptive Audio Processing
- Distinctive Dial Presence
- Professional Quality Finish
- Competitive Impressiveness
- Faster Audience Capture
- Higher Audience Loyalty

The Competitive Edge

- Faster Audience Capture
 - Longer Listener Retention
 - Higher Audience Loyalty
 - Higher Hit Ratings
 - Stronger Market Position
 - Greater Sales Volume
 - Longer Client Retention
 - Greater Profitability
- and...
- A Greater Assurance of “Mission Accomplished”

18.2 Basics of Processing

18.2.1 Leveling

Audio programs contain a wide range of levels over time. It is usually desirable to pull all the program elements, voices, music, commercials, etc., together toward a consistent sound level. This makes listening to the program much more enjoyable without the need to constantly readjust the volume control. Bringing together the sound levels of a program is called “Leveling”.

18.2.2 Peak Limiting

The peaks of a program don’t necessarily relate directly to the sound level as our ears detect it. If the leveling was done to satisfy the ear, then there remains a great deal of peak variability in the audio stream. This can cause problems in trying to get a full level of performance from the streaming audio encoder. It is therefore desirable, in addition to leveling the program, to cause the peak audio levels to be limited to a maximum value, and preferably at relatively lower values than they originally occur. This is because the analog to digital conversion and the subsequent digital data compression algorithms all relate to the peak level of the audio stream. If the audio peaks are processed to be consistent and predictable in level, then the streaming audio encoder can be driven to full scale encoding without ever having an overload occur. There will be no need to allow excessive headroom for occasional high peak levels. This results in better sound encoding at all bit rates.

18.2.3 Designing the Sound

Beyond the technical reasons for using audio processing, there is the notion of aesthetics. Some may relate this to “competitiveness”. The worldwide web has an ever-growing abundance of streaming audio sources to choose from. With such a great amount of competition between webcasters, it can be important to “stand out on the dial”. You may want to design your sound to be something different from all the others, and hopefully sound impressive to listeners. Unfortunately, CD’s and talk played unprocessed and raw may sound great at the time they are recorded, but usually sound low in level and unprofessional when broadcast over the internet media. With good audio processing you can obtain a professional finish that listeners will appreciate, leading to greater audience loyalty.

18.3 Getting the Best Processing

You could piece together a combination of various compressors, limiters and equalizers in an effort to build a complete audio processing system. This is seldom satisfactory, since the equipment that is designed for general studio applications is not specialized for broadcast processing. You will not be able to keep peaks under control without creating a great deal of sonic artifacts such as pumping and hole punching. The Model 2020 melds many specially designed audio processing techniques, including at least 11 that are patented exclusively

by Aphex, into one unit. The entire complex processing chain works interactively and harmoniously, specifically avoiding all the sonic aberrations caused by conventional audio processors.

18.4 Digital Versus Analog Processing

The output media of web broadcasting is of a digital nature. Because of that, some people think it makes the most sense to process the sound only in the digital domain. Why, then, did we build the Model 2020 in the analog domain? The fact is that digital audio processing is very limited in its capabilities, especially sound quality. This is because digital audio is time sampled and quantized. Digital audio processors have to contend with many, many approximations within their mathematical algorithms and inevitably generate digital distortions from truncation, rounding, and aliasing. That is why digitally processed sound is edgy and dry. It may have an impressive zippy or splashy quality at first listen, but that effect quickly becomes irritating. Analog processing, especially with the advanced circuits of the Model 2020, is inherently free of these aberrations. To provide the means to interface with a digital audio system, we included an optional 20-Bit AES/EBU input/output module. The digital input is converted to analog, processed, then converted back to digital. Any small quality loss that may be experienced through the converters is overshadowed by the greatly improved sound quality of the analog processing.

18.5 How to Use the Model 2020

Generally, you will interface the Model 2020 between the audio source and the streaming media encoder. When the encoder is an external hardware box, that becomes relatively easy because you can gain direct access to the encoder's analog or digital audio input. However, if the encoder is in software, then you will need to interface with the computer's audio board input cabling. The idea is to get as close as possible to the encoder's input point.

18.5.1 Set the Input Level

Once the audio source is fed into the Model 2020, you need to normalize the input gain. Set the Input Gain so the input level meters are peaking at 0 for a nominal program level. You operate the input gain through the Processing Input/Output menu page.

18.5.2 Set the Output Level

If you are using the digital output, then you don't need to worry about this. The Model 2020 automatically normalizes the output peaks to 1dB under digital full scale. If you are using the analog outputs, then you need to set the output level to properly drive your encoder's input. You operate the output gain also through the Processing Input/Output menu page. Your encoder may have an input meter to indicate the proper drive level, but if not, set the output to the highest level that still gives undistorted sound when listening to the stream through your reference media player.

18.5.3 Set Up the Processing

Now is the time to go to the chapters on setting up processing. First you will set up all the Globals and save them. Then, you will experiment with the presets. Finally, you will fine tune your own sound. This process may take anywhere from 10 minutes to a couple of hours depending upon your dedication and level of satisfaction.

18.6 Bit Rate Effects

Web audio media varies through a wide range of bit rates and consequential audio quality. Very low bit rate encoders often benefit from limiting the audio bandwidth of the input signal. In many cases, you cannot anticipate what bit rate will be served for any given stream. Nevertheless, you have only one audio processor and it needs to sound great on the highest bit rate or why bother. Therefore, it is not possible to specifically optimize the audio processing for low bit rate and high bit rate at the same time. We have found, however, that the Model 2020, if adjusted to sound great on a high rate stream such as 22KB, it also sounds quite good at low rates like 3KB. If you have separate high and low rate converter banks, then you can aid the low rate converter by adding a lowpass filter such as a parametric equalizer shelved off at 5-6KHz, to the converter's audio input. This may improve any splatter distortion you may be experiencing. It should be emphasized that splatter is not a problem caused by the Model 2020, rather it is a characteristic generally inherent in all low bit rate converters.

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