CIRCUIT CELLAR ONLINE

FEATURE ARTICLE

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MPEG and DSP Integration

For their final project at the University of Calgary, Priyesh, David, and their team members wanted to demonstrate MPEG decoding on Analog Devices' SHARC 21061. Here, they share what they learned about MPEG history, MP3, and the music industry's future.

A final-year team project in an electrical engineering program can be both fun and frustrating. Fun because you and your team get a chance to work on something major of your choosing. The frustration is par for the course but good training for the experiences found in many projects, whether within the university or the real world. There are time constraints conflicting with unfulfilled expectations from yourself, your teammates, and your project manager.

For our project, we decided to learn enough about the MPEG protocol and DSP processors to demonstrate the concepts of MPEG decoding running on an Analog Devices' SHARC 21061 evaluation board. The project naturally broke off into a number of parts. One component adds additional memory to the SHARC board, and another component finds a suitable algorithm and ports it to the SHARC board. We tackled these components with the help of our teammates, Matthew Mastracci and Luigi Iuliano, and our faculty advisor, Dr. Mike Smith.

The first step was to understand something about MPEG itself. The five of us had heard of it, had used it, but what was it? This article is the result of our background research into MPEG. It will cover MPEG history, current uses, future perspectives, and MP3's affect on industry.

STARTING AT THE BEGINNING

The Motion Picture Experts Group (MPEG) audio compression algorithm is an International Organization for Standardization (ISO) standard for high-fidelity audio compression. [1] The MPEG standard is a high-complexity, high-compression, and high-audio-quality algorithm. [2] Digital compression allows more efficient storage and transmission of data, and the many forms of compression offer a range of encoder and decoder complexity.

Already, the MPEG standard has gone through a number of stages. These are illustrated in the timeline shown in Figure 1.

When MPEG began its work to develop a standard for digital compression, its goal was to develop an algorithm that could compress a video signal and then be able to play it back off a CD-ROM or over telephone lines at a low bit rate. The intention of the group was to achieve a quality level that could deliver full-motion full-

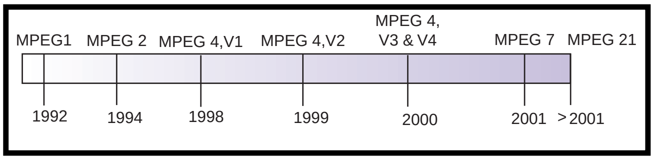

Figure 1—*A timeline showing the various MPEG standards dating from 1992 to the present and possible future standards.*

screen VHS quality from a variety of sources.

This initial standard was not broadcast quality, but it was good enough to display on a computer monitor or to playback from a consumer multimedia device. Because of the need for increased compression and better quality, a new standard was needed that would suit the purposes of the broadcast industry. MPEG started a second effort that is known as MPEG-2.

Along with the development of MPEG-2, work began on the MPEG-3 standard, which was directed towards the market of High-Definition Television (HDTV). MPEG-3 targeted HDTV applications with sampling dimensions up to 1920 × 1080 × 30 Hz and coded bit rates between 20 and 40 Mbps. However, research established that, after finding an optimal balance between sample rate and coded bit rate, MPEG-2 and MPEG-1 syntax could work well together for HDTV rate video. MPEG-3 no longer exists because HDTV became part of the MPEG-2 standard.

The development of MPEG-4 began in September of 1993 in Brussels, Belgium. This standard targeted low bit rate coding of audio-visual programs and required the development of fundamentally new algorithmic techniques. The sampling dimensions were up to 174 × 144 × 10 Hz with coded bit rates between 4800 and 64,000 Mbps. The MPEG-4 standard enables many new applications including interactive mobile multimedia communications, videophones, mobile audio-visual communication, remote sensing, interactive multimedia databases, games, interactive computer imagery, and sign language captioning.

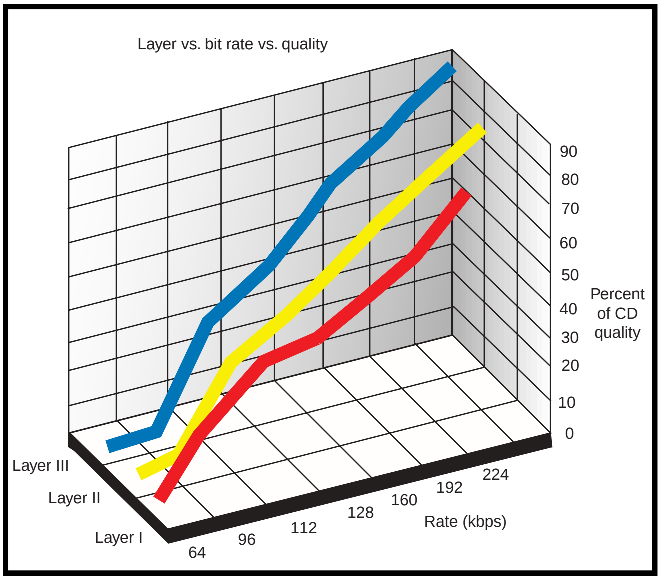

Figure 2—*A comparison showing Layers I, II, and III and the bit rate for each layer to achieve the percent of CD quality.*

MPEG AUDIO ENCODING AND DECODING

The MPEG audio standard has three distinct layers for compression. Layer I forms the most basic algorithm with minimal encoding and is best suited for bit rates above 128 kbps per channel. Layer II has an intermediate complexity and is targeted for bit rates around 128 kbps per channel. Possible applications for this layer include the coding of audio for Digital Audio Broadcasting.

Layer III is the most complex but offers the best audio quality, particularly for bit rates around 64 kbps per channel. This layer is well suited for audio transmission over ISDN. Figure 2 shows the minimum compression ratio needed to achieve 100% CD-quality with the different codecs or layers.

Layers II and III are enhancements that use some elements found in Layer I. Each successive layer improves the compression performance, however, it is at the cost of greater encoder and decoder complexity (see Table 1). At the same time, all three layers are simple enough to allow single-chip, real-time decoder implementations.

Layer	Compression
1	1:4
2	1:6...1:8
3	1:10...1:12

Table 1—*The maximum and minimum compression possible with each layer. Layer I, being the simplest, provides the least compression, and Layer III, having the most complex algorithm, provides the most compression.*

We will now describe MPEG-1 Layer III, which is a perceptual audio coding scheme that reads the audio signal. MPEG-1 Layer III is also known as the new MP3 standard. The algorithms in this layer are used for applying a psychoacoustic model to achieve a 12-fold compression rate. This model uses the properties of the human ear, trying to maintain the original sound quality as far as possible.

Figure 4 shows the basic modules associated with MP3 technology. You can see that there are two main components—the encoder and the decoder. The encoder takes a digital audio signal and compresses it into an MP3 file. It then sends the MP3 file through a channel, in this case the Internet, to the decoder. The decoder decompresses the MP3 file and converts it to a digital audio signal that can be played on a soundcard or any other audio device.

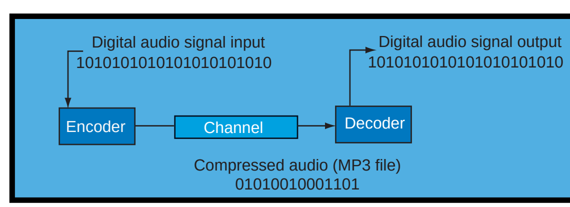

Figure 3—*Input is transferred to the output via a channel.*

ENCODER TECHNOLOGY

The encoder is a computer algorithm that uses psychoacoustic models, also known as perceptual models, to compress the raw digital audio file into an MP3 file (see Figure 4).

The compression algorithm uses psychoacoustic models to reduce the size of the raw audio file, a uniform quantizer, and encoding. Using a Fast Fourier Transform (FFT), the original audio is transformed from the time domain to the frequency domain to provide amplitude for every frequency component. [4]

The psychoacoustic model uses the fact that the human ear can only hear sounds of certain amplitudes and only frequencies between 15 and 20,000

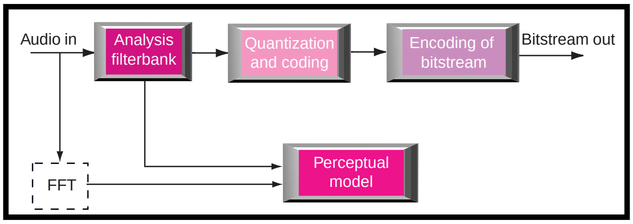

Figure 4—*Here you can see the various blocks of the encoder and how they interact with each other.*

Hz. Knowing this, the size of the audio signal is reduced.

Take, for example, an audio signal that has a loud bang (uses a large number of frequencies) and a soft sound (uses a small number of frequencies) at the same time. The compression algorithm eliminates the soft sound because it is unlikely that anyone would be able to hear it. This technique is called sound masking. If a strong signal appears, the weaker signal behind it is not perceivable. The MPEG algorithm removes this weaker signal (see Figure 5). This is advantageous because the information that is removed no longer needs to occupy hard disk space or Internet bandwidth.

The compression algorithm reduces the bandwidth of the original signal because, in most cases, frequencies at the extreme of our perception (i.e., 15 Hz and 20,000 Hz) cannot be heard, so they are filtered out. Although these examples are simplified, these basic ideas are used to compress the digital audio signal into the MP3 file format.

STREAMING THE MP3 FILE

One of the essential features of an MP3 file, after it has been encoded, is its ability to be "streamed." Streaming is the process of sending a segment of bits to a computer via the Internet. Streaming MP3 songs makes it possible to have CD-quality radio stations on the Internet.

With older sound file formats like WAV, AIFF, and AU, the sound files are large. To play these file types, you have to wait for the entire file to download over the Internet before you can start listening.

With streaming audio you can start listening right away. In other words, the sound starts playing without waiting for the entire sound file to finish downloading. Streaming audio is only possible with highly compressed sound file formats or fast modems.

In the past, most on-line radio stations sounded like an AM radio station because the signal's bandwidth had to be reduced in order to be streamed to a standard 56-kbps modem. Now, with the use of cable modem technology, these same radio stations can use MP3 technology to broadcast high-quality sound over the Internet.

DECODER TECHNOLOGY

The decoder is a computer algorithm that converts the MP3 file format to a WAV file capable of being played on a soundcard or other audio device (see Figure 6).

The decoder works by taking in a bitstream, using the reverse process of the encoder. First, the bitstream is decoded, then reduced, and finally, the Inverse Fast Fourier Transform (IFFT) is applied. The IFFT converts the signal back to the time domain, where it can then be played on a soundcard or other sound device.

COMPLEXITY OF THE DECODER

The decoder complexity is directly related to the type of file format it will decode. Like the encoder, the complexity increases as the layer increases, so a Layer I decoder is simpler than a Layer III decoder. To meet the MP3 decoder standard, a decoder must be backward compatible. Therefore, if a decoder states it decodes MPEG-1 Layer III, it must also decode Layer I and II. [2]

SOUND QUALITY

After the data is encoded, the decoder can decode a variety of sound qualities. These qualities can range from telephone sound to CD-quality (see Table 2).

Table 2 shows how sound quality and bit rate are both directly related to bandwidth. Most of the time, bandwidth is a limiting factor in a communication system. This is why the phone system sounds so poor. The sound you hear in the phone system uses just enough bandwidth to allow individual voices to be distinguished. [6]

IMPACT ON THE MUSIC INDUSTRY

With the creation of MP3 technology, the ability to store and retrieve audio files has increased dramatically. An average song in the traditional CD format is usually 60 MB. Most Internet users' connection speed is 56 kbps, which means it would take approximately 4.1 hours to download a song because the average download rate from an FTP server for a 56-kbps modem is 4 kbps. Most people consider this too long and not worth the effort. With MP3 technology and its 12:1 compression ratio, it is now possible to download the same song in 20 minutes. With a cable modem, the time is reduced even more, because the average download rate from an FTP server is 50 kbps, therefore, it only takes 1.5 min. to download the MP3 file.

The ability to download quickly has created a growing trend of users who download and trade MP3s on the Internet. There are many sites that link people to FTP servers, which carry illegal MP3s. An even stranger trend is that people are now using these sites to make money, but not from the MP3s themselves. Rather,

Sound quality	Bandwidth (kHz)	Mode	Bit rate (kbps)	Reduction ratio
Telephone sound	2.5	Mono	8	96:1
Better than shortwave	4.5	Mono	16	48:1
Better than AM	7.5	Mono	32	24:1
Similar to FM	11	Stereo	56…64	26…24:1
Near CD	15	Stereo	96	16:1
CD	>15	Stereo	112…128	14…12:1

Table 2—*The various sound mediums and the corresponding bandwidth, mode, bit rate, and reduction ratio.*

they get people to click on their ads on the page, and when all the ads are clicked, the user is allowed access to the FTP server with all the MP3s.

In other cases, people are using MP3s to increase the number of page views or hits, so they can charge advertisers more money to place advertisements on their page.

ILLEGAL MP3S

Is MP3 legal? In short, yes! MP3 technology is legal. However, MPEG.org states, “Note that downloading or streaming music from the Internet is illegal unless the copyright owner explicitly allows free downloads.” Therefore, if you have the permission of the copyright owner to download their music, you can do it without any fear of repercussions for your actions. [7]

The MP3 compression algorithm is legal because it is just an algorithm. The aspect of the technology that is illegal is its ability to compress copyrighted CD music and make the compressed version of the CD available on the Internet for people to download for free. The MP3 technology and faster Internet connection speeds make it even easier for people to download copyrighted CDs.

In a joint co-operation with Recording Industry Association of America (RIAA), more than 120 companies and organizations representing a broad spectrum of information technology and consumer electronics businesses, Internet service providers, security technology companies, and members of the worldwide recording industry have joined forces. Their weapon is known as Secure Digital Music Initiative (SDMI). It is hoped that the technology developed by SDMI will make it possible for legitimate buying and selling of digital music on the Internet. [8] BMG Entertainment, EMI Recorded Music, Sony Music Entertainment, Universal Music Group, Warner Music Group, RIAA, and International Federation of the Phonographic Industry are just a few of the companies taking action.

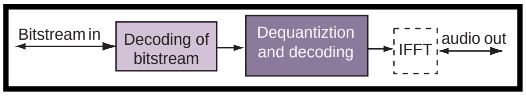

Figure 6—*Here you can see the various blocks of the decoder and how they interact with each other.*

SDMI’s work is based on the core principles that copyrights should be respected and those who opt to do so should be able to use unprotected formats. This flexible approach will enable a new market that works for consumers, artists, manufacturers, and content providers.

CURRENT USES

The MP3 compression algorithm offers the ability to store a vast amount of audio information in a small storage space. This means that whole CD libraries can be stored on several CDs rather than hundreds. It also means that CDs can be downloaded from the ’Net, sparking online companies such as MP3.COM, which markets MP3s online.

This technology also gives new artists a chance to distribute their music without a recording label. Most, if not all, free MP3s on the Internet are created by independent recording artists.

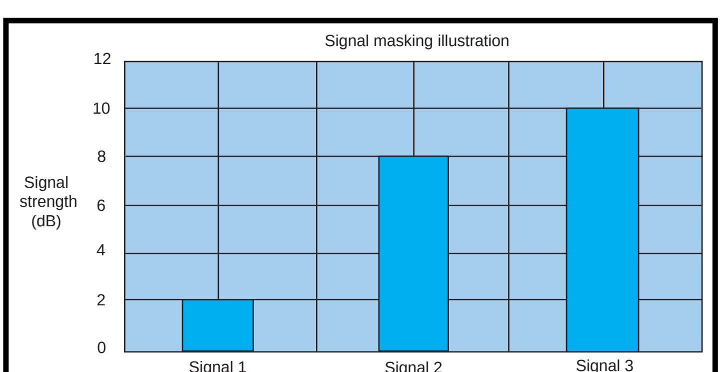

Figure 5—*Signal 1 and Signal 2 would be masked, and Signal 3 would be used by the MPEG algorithm.*

FUTURE USES

No one knows what the future holds, but everyone loves to speculate. The future could be bleak for MP3 technology, because technology constantly changes. However, it is unlikely that MP3 technology will completely die because of its widespread use. Before it dies out, it will probably change form becoming MPEG-1 Layer 4 or something similar. We will probably witness the next generation MP3 standard, which will be more sophisticated.

The best place for MP3 technology is in the car. With it, you can store your whole CD library on several CDs, making them easier to be played in the car. This means, you will need an in-car MP3 decoder. If these decoders become popular, the market for MP3s will continue expanding rapidly. Currently, there are several in development. With the growing amount of remote Internet access, it is also possible to have MP3 streamed directly into you car via cell phone Internet technology. You would be able to listen to any radio station around the world with quality sound in near real-time.

There is talk of using this technology to improve the sound quality of Internet Telephony. The only problem is that it is difficult to encode MP3 in real-time without hardware support. There are hardware-based MP3 encoders, but they are still not ready for the consumer market. Some of the other applications include a portable MP3 player, MP3 via cell phone, and many more. The list of possible applications is endless.

THE LAST NOTE

The MPEG-1 Layer III, or MP3 technology, is unique in that it uses perceptual models, rather than the classical digital bit compression to compress audio files. By using this technique, it is possible to compress audio files without losing audible signal quality.

MP3s offer society the ability to store high quantities of audio information in a relatively small amount of memory. It also offers many independent recording artists the opportunity to create and distribute their music inexpensively over the Internet. However, the most impressive part of this technology is what it can do in the future. With it, we may be able to listen to high-quality radio broadcasts from around the world in real-time from our automobiles.

In its short 12-year history, this technology has grown from obscurity to one of the most widely used and recognized audio standards ever developed.

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