Streaming Codec Study Report
For NPR Digital Media

DRAFT

John Kean

Ellyn G. Sheffield, PhD

NPR Labs

February 19, 2013

CONTENTS

INTRODUCTION AND EXECUTIVE SUMMARY

This report presents the results of research and testing in support of a new Internet audio streaming system to be launched by NPR Digital Media, with a focus on delivery over mobile wireless networks to smartphone and tablet devices using a custom player application. The objective of this study was to identify the best digital audio codec and streaming rate. Additionally, we explored the effects of audio level shifts on consumers' preferences and behavior and developed a solution to loudness management for playback in a variety of listening conditions.

This section provides an executive summary of the key findings and technical recommendations. The appendices discuss implementation, as well as explanations of the research and methodology.

CODEC CHOICE AND OPTIMAL BIT RATES

Digital audio codecs are the heart of a practical streaming system. They make Internet streaming commercially feasible by vastly reducing the bit rate required of the audio data without noticeably reducing the audio quality. Without them, streaming would require too much data bandwidth to be affordable.

Codecs vary in their ability to maintain sound quality at lower bit rates. The older ones, such as the familiar MP3 format, are the poorest at hiding audible "artifacts" that result as the bit rates are reduced. Through continual research and development, digital compression techniques have continued to evolve, leading to new codecs that provide high sound fidelity at ever-lower bit rates.

Besides saving on transmission expenses related to the data bandwidth, minimizing the bit rate has direct benefits for the listeners, because lower stream bit rates:

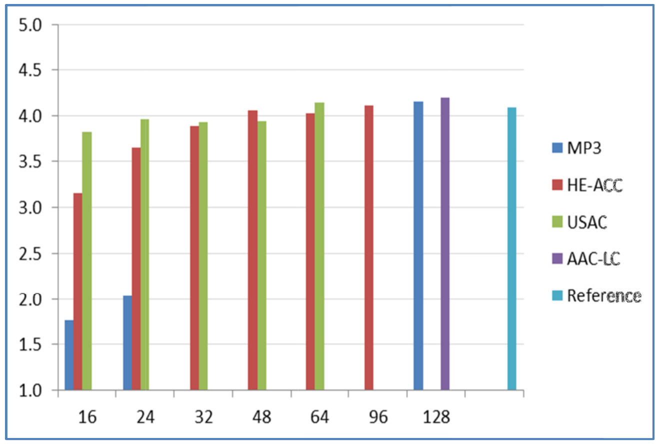
- Start faster when a stream is selected, much like a radio plays as soon as tuned in to a station;
- Experience markedly higher reliability (fewer dropouts) in mobile wireless networks, and
- Restart audio faster after a dropout occurs
- Save potential data charges for the listener.

This study included real-time field measurements of wireless mobile networks (see

Streaming Requirements/*Wireless Networks* in the appendix), which explains that as the bit rate rises, a rise in the frequency and duration of data interruptions ("dropouts") occur. In fact, data show the amount of interruption rises steadily at rates above approximately 32 kbps, whereas, at lesser rates, outages tend to be caused by weak cellular signals. Not surprisingly, testing in the Washington, DC area also showed that stream reliability was worst during morning and evening drive-time – the peak listening times for most public radio programming.

There are numerous codecs available for use by portable devices and computers. We tested six of the leading codecs with expert listeners over a wide range of bit rates (see the appendix for *Selection of Codecs by Well-Informed Listeners* for details).

After identifying the High Efficiency AAC (HE-AAC) codec as the top performing codec, consumers were tested to substantiate the expert listeners' recommendations (see the appendix *Audio Quality of Codecs - Consumer Test* for details). **Error! Reference source not found.** shows how consumers rated the HE-AAC codec, shown in dark blue, on a vertical scale of Mean Opinion Score ranging from 1 (worst) to 5 (best) for the music genre. Notice that there is relatively little difference between the overall (encoded/decoded) quality and the original reference audio, shown to the far right in light blue. Over a range of bit rates suitable for mobile streaming, our test data and industry research show good agreement between the listener scores with music and speech programs.



odecs

Recommendation 1: Of the codecs currently available, the standout, overall, is High Efficiency-Advanced Audio Coding ("HE-AAC", also known as AAC+ and aacPlus) for streaming. HE-AAC is preferred for the following reasons:

- Provides the sound quality of MP3 coding at less than half the bit rate;
- Commonly available in Apple iOS (all versions) and recent Android operating systems;
- Licensing fees paid by end-product manufacturers, and no fee to program originators;
- Support for metadata transport for loudness normalization and ancillary data functions.

As discussed fully in the appendix, the optimal bit rate must consider various aspects of the listener's experience, which must balance sound quality at higher bit rates and streaming reliability at lower bit rates. In our view, this balance is best met with the following considerations:

- **For both music and speech streamed with the HE-AAC codec, the optimal codec and bit rate is 48 kbps**;
- **For non-real time playback, such as podcasts, the optimal bit rate for music and speech with the HE-AAC codec is 64 kbps**;[1]
- Higher bit rates are unnecessary as listeners heard minimal, if any, improvement.
- Provide for updating the encoder as new versions are released (updates usually on the encoding side, where the heavier data processing occurs). The streaming software should also be designed to support the future replacement with a superior codec.

[1] The reason that a higher bit rate may be used for podcasts is that these are file transfers to the player, which bridge interruptions in the flow of data over the Internet. File transfers may be bursty and interrupted, but these effects are unnoticeable during file play back. Live streams use relatively small data buffers to permit faster startup and recovery from dropouts, but they require a more constant flow of data.

The HE-AAC codec is almost universally available in portable devices operated by iOS and Android (since version 3.1). There are earlier Android devices that may not have HE-AAC on board, however, as these devices are now more than two years old, this percentage is declining as consumers replace their devices with newer models having versions of Android OS with HE-AAC.

LOUDNESS MANAGEMENT WITHIN AND BETWEEN STREAMS

Uniformity of perceptual loudness is an important consideration for the listening experience. As discussed in detail in *Loudness Consistency/Current Experience*, the uniformity of public radio streams is quite poor in terms of loudness, audio bandwidth and levels of digital compression artifacts. The chart below shows a series of audio samples from 49 public radio stations, all carrying the same news program from NPR. The vertical position of the dot measures the subjective loudness on a decibel (dB) scale –a desirable range is ±3 dB. In this test, however, streams are varying over more than a 15 dB range in audio transmission level.

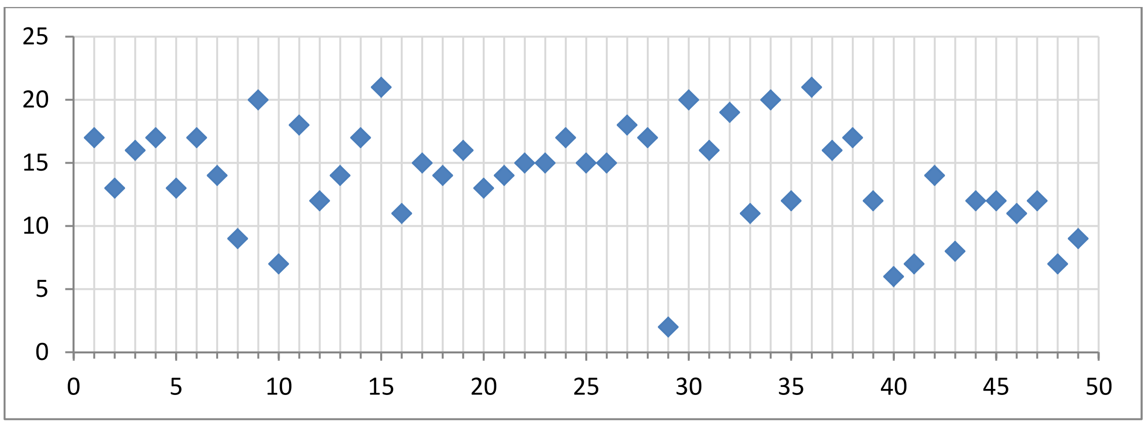


Figure 2 – Audio loudness of 49 public radio station streams carrying *Weekend Edition Saturday*

This project included a systematic study with consumers to determine their reaction to changes in loudness, as reported in *Loudness - Consumer Test 2*. For cases such as switching from stream to stream, or hearing one announcement at certain level followed by another announcement at a different level, we measured the amount of change in audio level that is annoying to listeners. **Findings indicate that the irregularity of audio levels between public radio streams exceed the tolerance of listeners. To ensure a pleasant listener experience, the transmission level of each stream should be normalized, based on its long-term loudness, as measured by the ITU standard loudness meter.**

The effect of loudness changes within a stream is important for listener satisfaction, depending on external environmental factors. For example, listeners on a noisy subway need tight control of dynamic range to hear speech or music programming clearly, but in better listening conditions, the natural dynamics of human speech or a musical performance can add to the experience and enjoyment of listeners.[2]

---

[2] As an experiment, we streamed a news/information and a classical music station for several weeks each with a direct connection to the console output, that is, no audio processing. Listener reports were universally favorable.

However, loudness variation should not be solved by more compression of dynamic range. We evaluated the dynamic range of many public radio streams and found that many use heavy compression and limiting, even with fine-arts programming. This may be suitable for broadcast transmissions which are subject to signal noise and interference, and which must conform to tight modulation limits. Protective audio processing is not required for Internet audio streams: it is a digital audio transmission system capable of more than 90 dB of dynamic range, which does not require dynamic range compression *per se*. Reduction in the dynamic range is useful for noisier listening environments, or simply to comply with the listener's taste. However, it is preferable to put this choice under the control of the listener, rather than a restriction that is universally applied to all listeners.

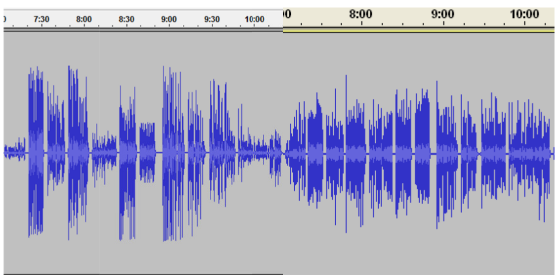


Figure 3 - 11 public radio streams as received (left) and after processing in an iPhone loudness control app (right)

We recommend a break from the tradition of dynamic compression at the transmission end, which forces a "one size fits all" treatment to audio. Figure 3 illustrates how active loudness management can be applied at the player. The thumbnail picture on the left shows samples from 11 public radio streams (20 to 30 seconds each) as they were received on an iPhone 4S. Considerable variation in level is evident in the heights of the streams - as much as 15 dB between adjacent streams. The picture to the right shows the same streams after activating active loudness management software that was added to the iPhone. The software was effective at raising the low level streams and bringing down the level of the loud streams. The audible effect was a more pleasant evenness in sound across the group, while each sample remained fairly true to its original form.[3]

Taking a cue from industry work on loudness management for high-definition television, we recommend the following:

- **Shift active loudness management to the stream player software**. This resolves the problem of audio control, both within the program and from stream to stream. Smartphones and PCs are capable of controlling the audio range, and there are already music player "apps" for smartphones that do this. This would put the control in the hands of the listener, according their needs and tastes.
- **Adopt loudness metering for production, setup and stream monitoring**. Measure the overall loudness and dynamic range of programs with ITU-R BS.1770-2 metering (see Appendix for details) to ensure levels "sound right" and are consistent with other programming.
- **Quasi-peak metering should not be relied upon as the only measurement procedure for audio stream level**. As demonstrated in the Appendix, peak measurement does not correlate well with subjective loudness.
- **Discontinue peak normalization of audio files in production**. This effort should be replaced by loudness metering, either manually or automatically, plus active loudness management in the player.

[3] This was a rather extreme example, as we would not expect to use this much control for streams that are transmitted near the proper level.

- **Discontinue persistent compression of audio dynamics before stream transmission**. With proper setup of transmission levels at the source, loudness management can be accomplished at the player, which also helps ensure that streams will sound better and more uniform to the listeners.

Measurement and control of audio transmission levels for public radio streaming is arguably a bigger challenge than selecting the codec and bit rate. Fortunately, we now have the technology to manage loudness effectively, and comfortably, for every listener.

ENCODING COMPUTER

As discussed in the Appendix, NPR Digital Media is supplying a server computer to stations for stream encoding, along with an audio interface and software. A Creative® model E-MU-0404 audio card was selected for the computer, a consumer product with high quality, but inputs that are not compatible with most stations' analog or digital audio plants. To minimize installation and operational issues, NPR Labs developed a custom audio interface box that ensures full compatibility and simple setup. The interface supports both analog and digital audio and has been tested in a live stream for several weeks at two member stations.

The E-MU 0404 audio card is discontinued and there is no direct replacement from Creative. We recommend a study to identify a suitable replacement. This should consider USB-connected audio interfaces, which are becoming common in audio systems.

Appendices

AUDIO QUALITY TESTS

This appendix details three tests conducted for Digital Media over the spring and summer of 2012. The first, *Selection of Codecs by Well-Informed Listeners* was conducted at NPR and at the Public Radio Engineering Conference in Las Vegas. The second and third *Audio Quality of Codecs - Consumer Test 1* and *Loudness - Consumer Test 2*, were conducted at Towson University with individuals from Towson and the surrounding Baltimore community.

SELECTION OF CODECS BY WELL-INFORMED LISTENERS

The purpose of this pre-test was to (a) critically evaluate several codecs that are currently being used or potential contenders for transmission over the internet, and (b) identify two codecs that would be presented to consumers in a final round of audio quality testing.

AUDIO CODECS FOR TEST

The codecs for this test were chosen to represent some of the most popular, as well as the newest (and arguably, best) available. The choices are listed in the table below, along with information on the relevant standards, if applicable, their history and licensing requirements.

Codec	Source Information	Description
LAME	lame.sourceforge.net	A free software codec using the MP3 format. Although a widely available download, it includes technology covered by patents owned by the Fraunhofer Institute. Its developers consider it educational software that, as such, does not infringe patents. Commercial use may require license fees.
MPEG-2 Layer III ("MP3")	ISO/IEC 11172-3 (1993) ISO/IEC 13818-3 (1995) http://www.iis.fraunhofer.de/en/bf/amm/produkte/audiocodec/audiocodecs/mp3.html	A widely-used codec used for consumer audio storage and transmission encoding and playback. It is commonly included in the firmware of most digital audio players. With some exceptions, the software is subject to royalty payments for use in broadcast, streaming and physical storage (see mp3licensing.com).
AAC-Low Complexity ("AAC-LC")	ISO/IEC 13818-7 (1997) ISO/IEC 14496-3 (2001) http://www.iis.fraunhofer.de/en/bf/amm/produkte/audiocodec/audiocodecs/aaclc.html	Advanced Audio Coding was designed as the successor to the MP3 format, achieving better sound quality at similar bit rates. It was originally developed in the mid-1990's and became an MPEG-2 standard in 1994. The Low Complexity ("LC") profile is intended for consumer applications, and because of its use in Apple's *iTunes*, is part of all Apple operating systems for iPhone, iPod and iPad device. It became a regular part of the Android operating system beginning with v3.1 (http://developer.android.com/guide/appendix/media-

		formats.html). Based on EBU testing (EBU Tech 3324, Sept. 2007), this codec is provides "good" to "excellent" quality at 128 kbps. Through its standardization in MPEG-4 the AAC family of codec often appears with that identification. The licensing model for the AAC family of codecs puts the fee on manufacturers or developers of end-user products, rather than users who broadcast or stream the audio (www.vialicensing.com).
High-Efficiency AAC ("HE-AAC")	ISO/IEC 14496-3 (2003 and 2005) http://www.iis.fraunhofer.de/en/bf/amm/produkte/audiocodec/audiocodecs/heaac.html	The addition of Spectral Band Replication (SBR) to the AAC-LC tool results in the High Efficiency AAC profile, known as HE-AAC or "aacPlus". With HE-AAC, the lower part of the audio spectrum in coded with AAC-LC, while SBR encodes a "guided recreation" the upper frequency spectrum. At rates below 128 kbps HE-AAC offers a reduction of up to 30% compared to AAC-LC. At bit rates of 24 to 32kbps, HE-AAC automatically adds Parametric Stereo processing, which extracts positional information from the stereo signal along with a mono down mix, to achieve a reconstruction of the decoded signal with the stereo image. This achieves a more efficient transmission than two-channel HE-AAC with similar overall scores for quality. When SBR alone is employed, it is referred to as "HE-AAC v1"; when Parametric Stereo is added, it is referred to as "HE-AAC v2" or "aacPlus v2". The version of HE-AAC in our testing can be determined by the bit rate employed for each test.
G.722.2 ("AMR-WB+")	ITU-T G.722.2	Adaptive Multi-Rate Wideband (AMR-WB) is codified at G.722.2, and ITU-T standard speech codec, or "vocoder". As a standard for mobile phones AMR-WB is intended to provide acceptable speech quality at very low bit rates, and multimedia support at moderate bit rates. Voice Age Corporation is a major developer of digital speech and audio compression technology for communications applications, which developed a "hi-fi speech and audio codec" version of its AMR-WB codec for both speech and music. This performance made AMR-WB+ a candidate for comparison. The technology is licensed to product integrators by VoiceAge.
Unified Speech & Audio Coded ("USAC")	ISO/IEC DIS 23003-3 MPEG-D Part 3 ISO/IEC 14496-3:2009/Amd 3 (2012) http://www.iis.fraunhofer.de/content/dam/iis/en/dokumente/pr/2008/voiceage_iis_engl.pdf	The Unified Speech and Audio Coded is a collaboration of Fraunhofer and Voice Age, which combines the technology of both companies to further advance the performance and efficiency of lossy digital compression. It uses time-domain vocoding tools for speech segments and SBR and PS coding techniques for music segments, and is able to switch or blend between the techniques dynamically in response to the program signal. It was developed by the MPEG as an MPEG-4 audio object type.

Other codecs were considered for the expert listener testing, however, the general requirement that the codec be available in the Android and iOS operating systems, and across a variety of consumer products limited the field. This requirement was overlooked in the AMR-WB+ and USAC codecs because of their deep standardization, which can help adoption in consumer devices. Indeed, the performance of the USAC, described elsewhere in this section, suggests it to be a future upgrade for NPR, once it becomes commonly available in products – or if no licensing fee is involved to bundle it in a player.

Another codec not available at the time of this study is Fraunhofer's Extended HE-AAC ("xHE-AAC"), which is the latest upgrade to the AAC family. Similar to the USAC, the xHE-AAC combines the advantages of speech and audio codecs, which makes it possible to outperform either coding approach alone. The Fraunhofer literature indicates increases in efficiency that lower the necessary bit rate for music of "good" quality by approximately 8 kbps for music and 20 kbps for speech, compared to HE-AAC v2. This would be another coder to consider for future implementation.

PARTICIPANTS AND METHODOLOGY

Eleven listeners (4 females and 7 males), all associated with NPR corporate or NPR-affiliates, listened to audio samples trial-by-trial to rank order the audio output of the codecs. Listeners were not taken from the general public for this effort. They were all well acquainted with codecs and the artifacts that codecs create. Within one sample condition (e.g., a female speech sample) participants listened to the same sample produced through six codecs, side-by-side, allowing them to compare small differences and rank order the codecs on a 0-100 scale. Figure 4 depicts one listener's data for one trial.

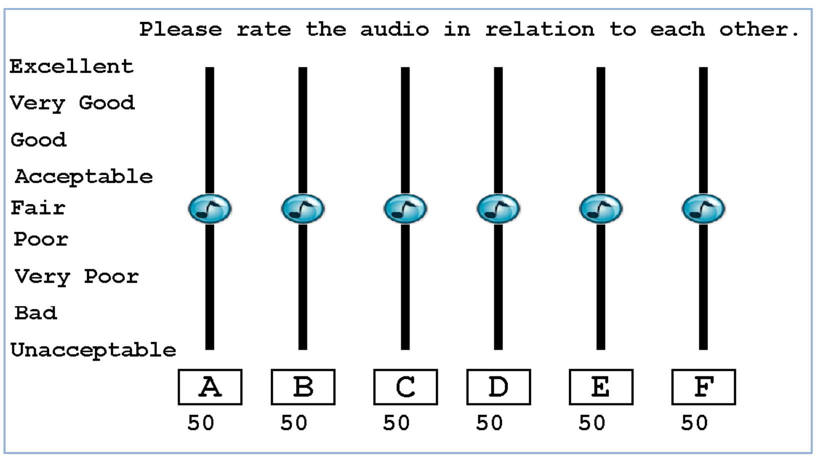


Figure 4 - Snapshot of MUSHRA screen

For each new trial, listeners heard randomly ordered samples, such that different codecs were placed on different slider bars. Selections included female and male speech, orchestra, single instrument, multiple voices and voiceover (i.e., commercials). The codecs, described above, were tested at the MPEG standard bit rates of 16, 32, 48, 64 and 96 kbps. While tests could be conducted at other bit rates, for example, to provide a more geometric distribution, some codecs are optimized for the MPEG rates. The selected rates were chosen to avoid the possibility of uneven test results.

SOUND SAMPLE MATERIAL AND PREPARATION

Audio material was selected from a variety of sources representing male and female speech, speech over a music bed (voiceover), singing, individual instruments and orchestral music. All of the material was produced from uncompressed PCM files with a sample rate of 44.1 kHz in 16-bit integer format. (Some music files were collected as 48 kHz recordings, which were converted to 44.1 kHz using Audacity 2.0 rate converter in the high-quality mode.)

All of the test material heard was played back as a WAV file. This required each audio sample to be encoded and decoded (transcoded) back to a WAV file for each codec type. The LAME samples were encoded and decoded in Reaper v4.26 using the version 3.98 codec. The AMR-WB+ transcoding was accomplished with the VoiceAge Evaluation Executable (http://www.voiceage.com/amrwbplus_eval.php), a Windows software tool that produces the output WAV file in one step. The MP3, AAC (including HE-AAC), and USAC files were furnished to the Fraunhofer Institute for transcoding.

To ensure that the listeners' scores were not biased by differences in loudness, the audio level of the samples were measured using the Orban Loudness Meter software, v 2.0.3 (downloaded from http://orban.com/meter/. Before transcoding, the audio gain of each WAV file was trimmed in Audacity 2.0 to approximately -23 dB LKFS, according to the "Integrated" indicator of the software tool's ITU-R BS.1770-2 loudness meter.

PLAYBACK CONDITIONS

The expert listener test was conducted with Sennheiser HD-600 headphones, connected to a Tascam US-144MKII audio interface, providing the audio connection from the test computer. Approximately half the test subjects listened in NPR Studio 5A, which provided a very quiet environment. The remainder of the listeners volunteered during the 2012 Public Radio Engineering Conference in Las Vegas, and tests were conducted in a quiet hotel room using the same headphones and headphone driver. A photo of the hotel listening room setup is shown in Figure 5. Listeners operated their codec test through E-Prime®, a PC program to automate the playback of codec audio test samples and collect the listener's responses.

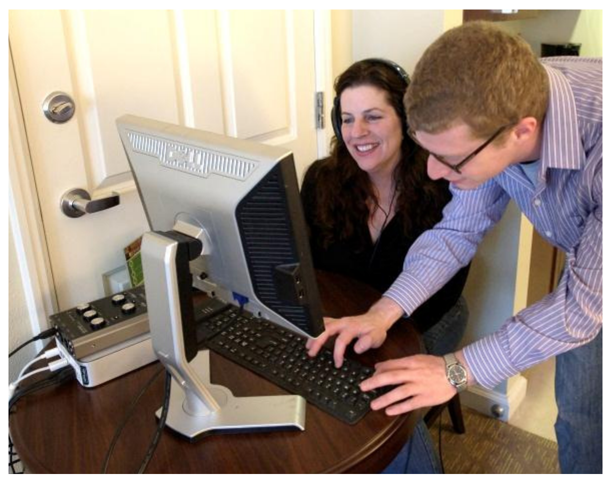


Figure 5 - Participant being set up for expert listener test

RESULTS

As can be seen from Figure 6, when all genres were combined, listeners felt that three codecs were satisfactory at 32, 48 and 64 kbps: USAC, HEAAC-V2, and AMR-WB+. Although LAME, Fraunhofer MP3 and AAC-LC were satisfactory at 96 kbps, they were significantly lower at all other bit rates, rendering them unacceptable as candidate choices for the consumer test.

Figure 6 through Figure 11 show results by genre. Notice that in speech, AMR-WB+ falls off precipitously at 64 kbps, highlighting its intended use for lower bit rates. With regard to music (i.e., orchestral, choral and single instruments) USAC and HE-AAC are consistently higher, with listeners ranking them as satisfactory starting at 32 kbps. (The data line for HE-AAC excludes 64 and 96 kbps data as some of that audio material was later found to have been inadvertently impaired in preparation for testing. However tests conducted by others suggest that the results for HE-AAC would have been similar to AAC-LC at 96 kbps.)

Interestingly, results were quite flat starting at 32 kbps, suggesting that listeners were mostly satisfied at lower bit rates and did not hear marked improvements at higher bit rates. Given these results, the audio quality consumer test included USAC and HE-AAC as the codecs of interest.

Figure 6 - Expert listener scores for ALL GENRES combined

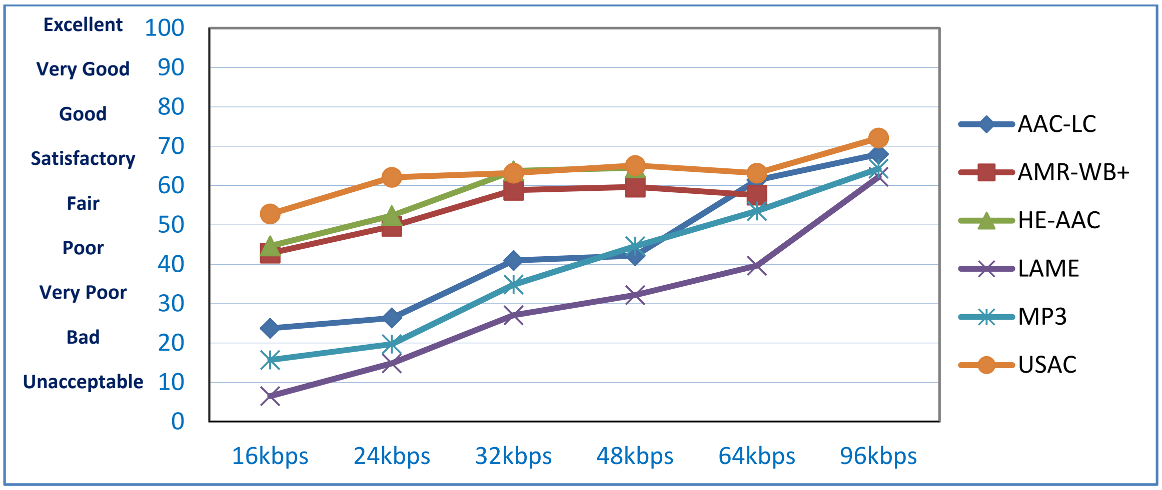


Figure 7 - Expert listener scores for SPEECH genre

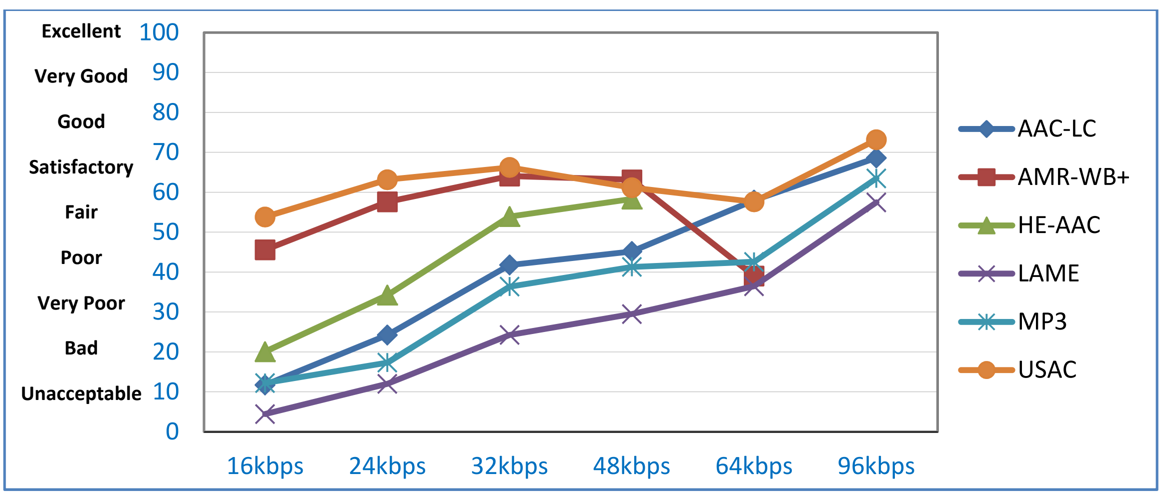


Figure 8 - Expert listener scores for ORCHESTRAL genre

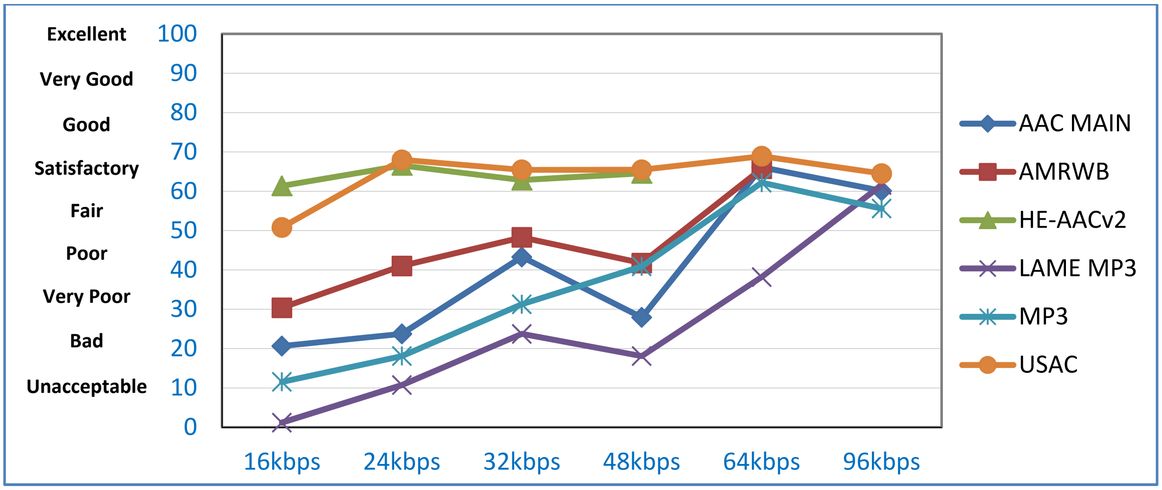


Figure 9 - Expert listener scores for SINGLE INSTRUMENT genre

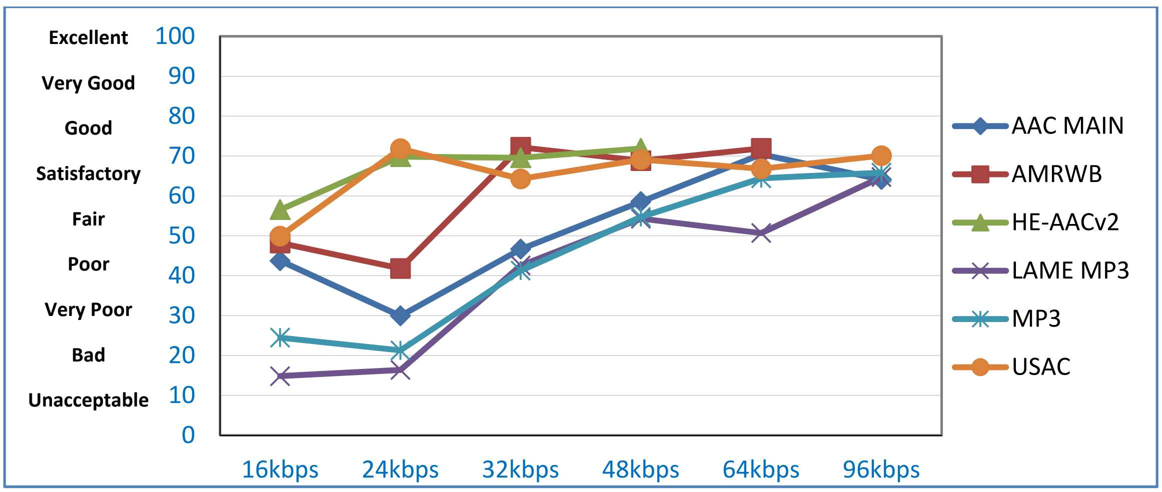


Figure 10 - Expert listener scores for VOICE OVER genre

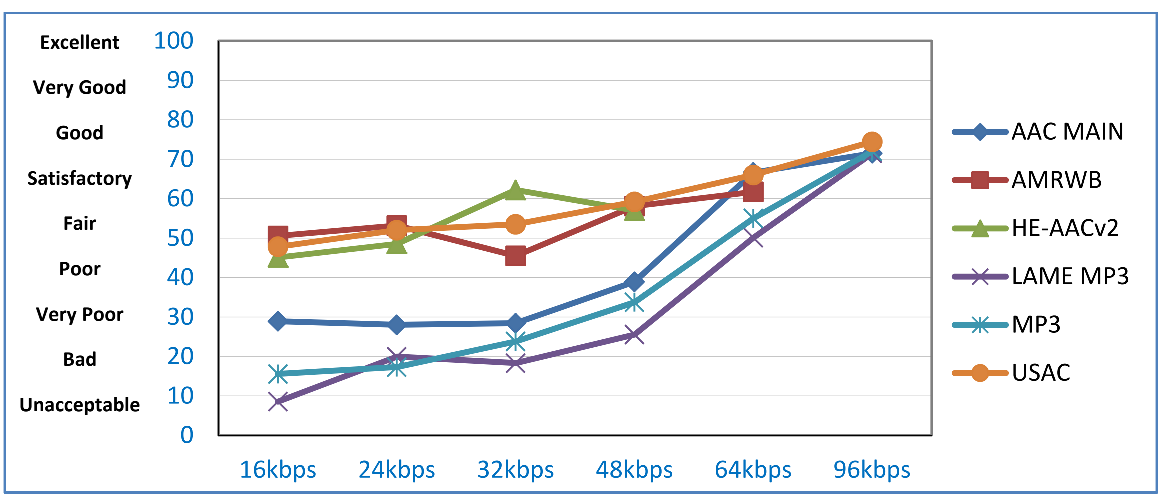
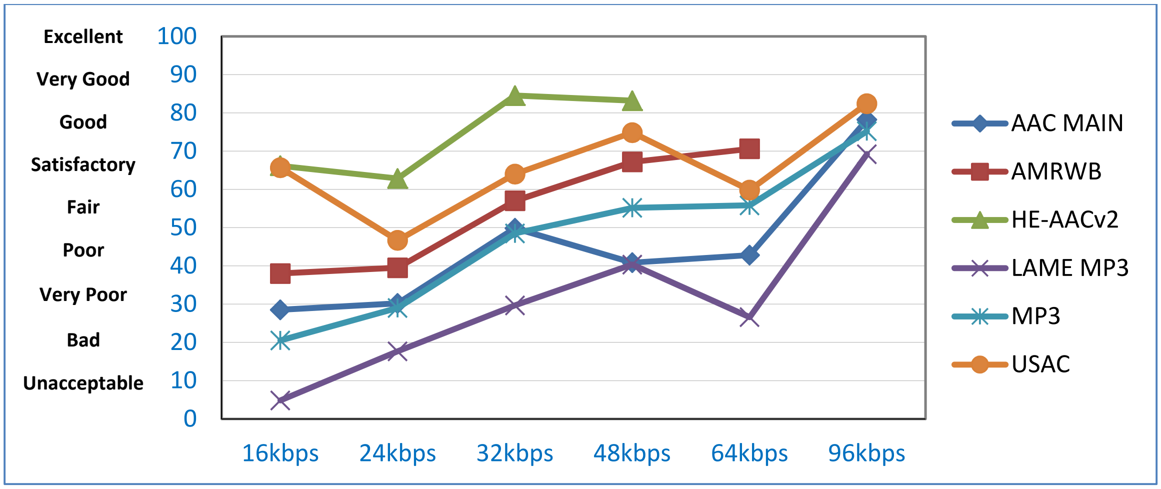


Figure 11 - Expert listener scores for CHORAL-MULTIPLE VOICES genre



AUDIO QUALITY OF CODECS - CONSUMER TEST 1

The purpose of this test was to determine how *consumers* would respond to audio transmitted at different bit-rates using our selected codecs - HE-AAC and USAC - and whether they would continue to listen to the transmission as the bit rate decreased. Consumers heard HE-AAC and USAC samples coded at 16, 24, 32, 48, 64 and 96 kbps, and heard samples coded with MP3 at 16, 24 kbps (low anchors) and 128 kbps (frequently used in streaming), plus AAC-LC at 128 kbps and an unprocessed sample, known as the "reference". The consumer test also presented the material under conditions closer to that of average listeners: the listeners used Apple® ear buds, rather than high-end headphones, and a Polk® iSonic© table radio. Also, the listening room was a small interior office with a moderate level of air handling noise. These tests were designed to determine the bit rate parameters for an Internet audio system, less one that is indistinguishable from the source under the most discerning of conditions than one that is capable of delivering a quality experience for all listeners. We believe the two tests inform that determination.

AUDIO CODECS UNDER TEST

The high scores of the HE-AAC and USAC codecs were primary considerations for the test. The AMR-WB+ codec performed well, too, but it is not common to the operating systems in current smartphones and personal computers. The same could be said for the USAC, but we feel this codec is on the best track for future adoption by smartphone manufacturers. (It also happens to be a joint development of Fraunhofer Institute and VoiceAge Corporation, the companies that were active in the development of the HE-AAC and AMR-WB+ codecs, respectively.) The MP3 codec was included to (1) provide a comparison to a codec and bit rate frequently used in Internet streaming (MP3 at 128 kbps), and (2) by operating at a low bit provide a low reference for the tests. The audio material for all three codecs was prepared in the same manner described for the expert listener tests and was transcoded at Fraunhofer Institute.

PARTICIPANTS AND METHODOLOGY

Forty participants were recruited from Towson University staff and administration via the university's "Daily Digest" announcement forum. See below for demographic details.

Age Group	Female	Male
18-30	5	5
30-40	5	5
40-50	5	5
50-65	5	5

Participants listened to a total of 250 speech and music audio samples, one-by-one. After each sample, participants were asked to answer 3 questions: (a) on a scale of 1-5 (bad, poor, fair, good, excellent), rate the audio quality for the sample; (b) on a yes/no basis, were you satisfied with the audio quality; and (c) would you keep listening to the radio or turn it off, given the audio quality of the sample. In two sessions, participants listened over loud speakers and ear buds. Sessions were counterbalanced such that 20 people were assigned to listening with speakers first and 20 people were assigned to listening with ear buds first. Participants were encouraged to adjust the volume to a comfortable setting before beginning the session, after which the volume remained constant.

RESULTS

DATA NORMALIZATION:

Within each genre, unique audio selections were presented for listening at different bit-rates. This methodology was specifically constructed in order to minimize the effect of listeners becoming overly fatigued by hearing the same selection too many times. However, for each audio sample, a non-processed reference was also included for listeners to rate. In order to ensure that the audio content for each selection was not unduly influencing people's ratings (Mean Opinion Score and On/Off decision), scores were normalized to their corresponding references. For example, if the MOS for a particular reference sample was 4.5, scores for codecs were calculated in relation to the 4.5; if the next reference sample was 4.2, those scores were calculated in relation to 4.2.

ANALYSIS OF GENDER AND AGE

A 2 (Gender: male, female) x 4 (Age group: 18-29; 30-39; 40-49; 50-65) Analysis of Variance (ANOVA) was conducted to explore whether men and women differently answered whether they would leave the radio on, given the quality of the audio. There was a main effect of age, $F(3,10920) = 8.14$, $p<.01$, showing that 18-29 year old listeners in general were more motivated to leave the radio on, particularly at higher bit rates. There was no substantial difference between male and female listeners.

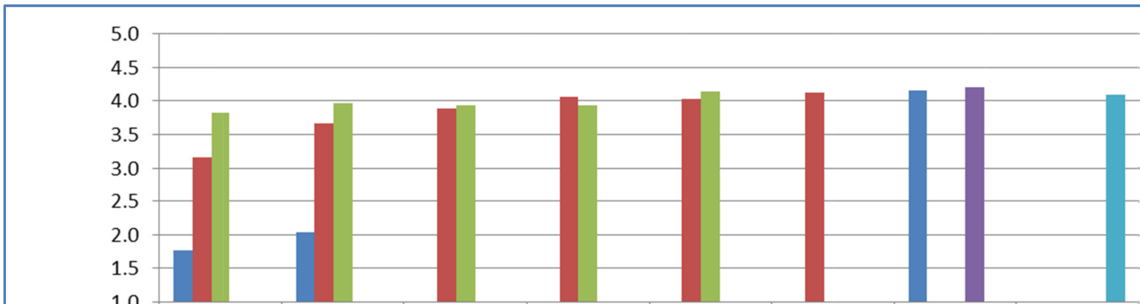
ANALYSIS OF EARBUD AND LOUDSPEAKER LISTENING

In order to determine whether consumers rated audio differently when listening over ear buds and loudspeakers, a multivariate analysis of variance (MANOVA) was conducted with type of listening, bit rate and genre as independent variables, and MOS, turn-off score, and satisfaction score as dependent variables. This analysis revealed significant differences for genre and bit rate on all measures but did not reveal differences for loud speaker or ear bud listening. Thus for the main analysis, all responses whether collected on ear buds or loudspeakers, were collapsed into one measure.

ANALYSES OF MOS, COMBINED AND SEPARATED BY GENRE

Figure 12shows listeners mean opinion scores when all audio selections were combined. Asterisk (*) indicate that the codec audio quality is statistically significant different from the Reference. Note that MP3 was significantly worse than HE-AAC and USAC at the lowest bit rates, and that at 32 kbps and above, HE-AAC and USAC performed extremely well, with listeners rating them as "good". Listeners did not distinguish between the audio quality of HE-AAC and USAC at 64 kbps and the uncoded reference sample.

Figure 12 - MOS - all audio combined



	16	24	32	48	64	96	128	Reference
MP3	1.8 *	2.0 *					4.2	
HEACC	3.2 *	3.7 *	3.9	4.1	4.0	4.1		
USAC	3.8 *	4.0	3.9	3.9	4.1			
AACLC							4.2	
Reference								4.1

Figure 13 - MOS - music

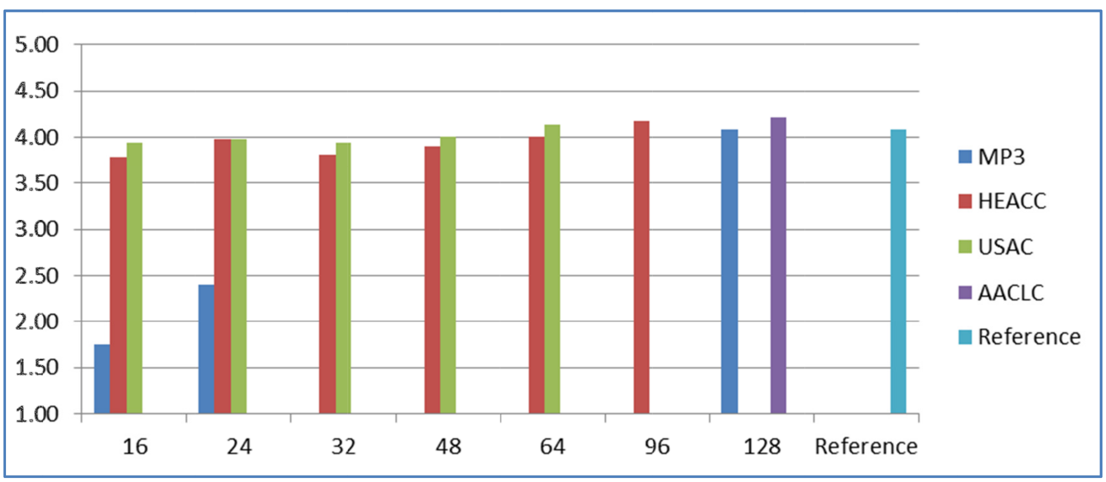


Figure 14 - MOS - speech

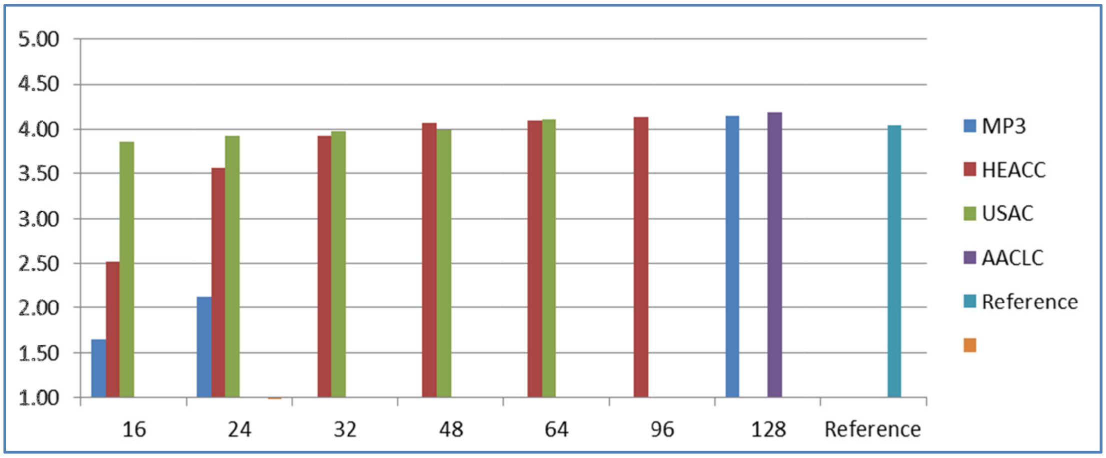


Figure 15 MOS - voiceover

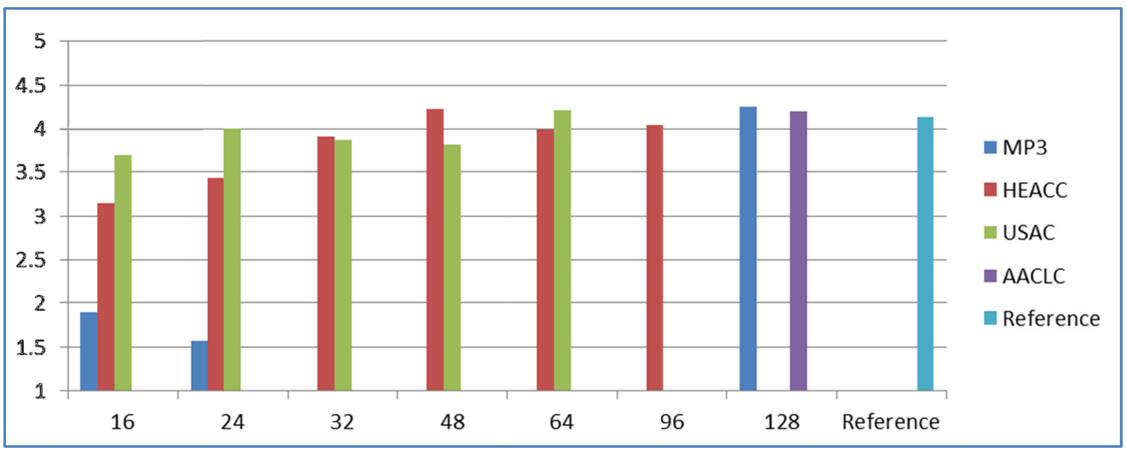


Figure 16 shows the On-Off scores with combined audio selections. When answering this question, 99% of listeners felt that they would leave the radio on at 32kbps and above. Even at 24 kbps, approximately 94% of listeners claimed they would leave HE-AAC on. Figures 12-14 show on-off scores by genre.

Figure 16 - Leave on - all audio combined

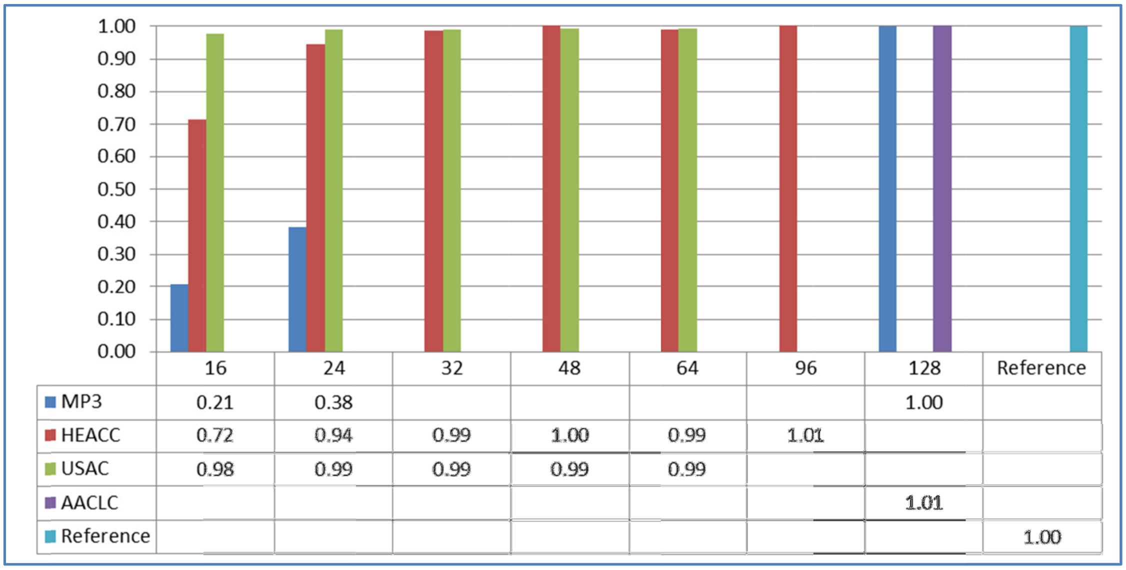


Figure 17 - Leave on - music

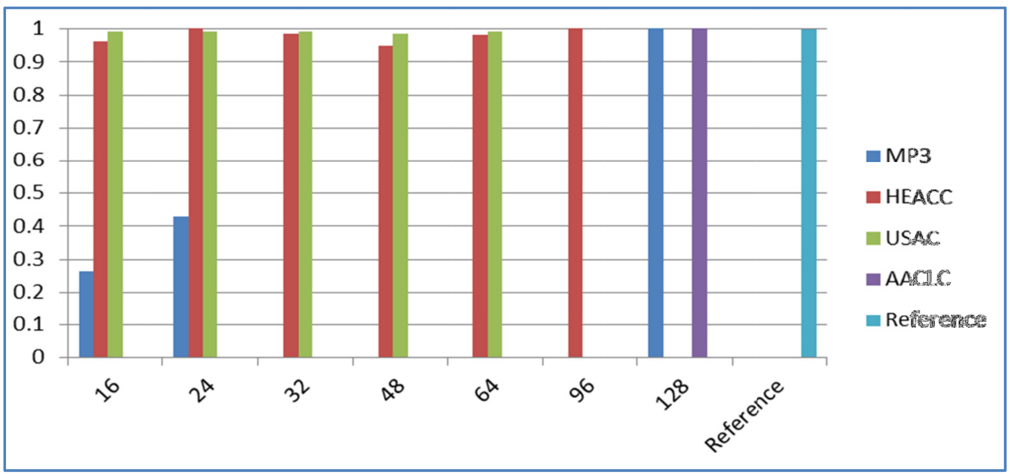


Figure 18 - Leave on - speech

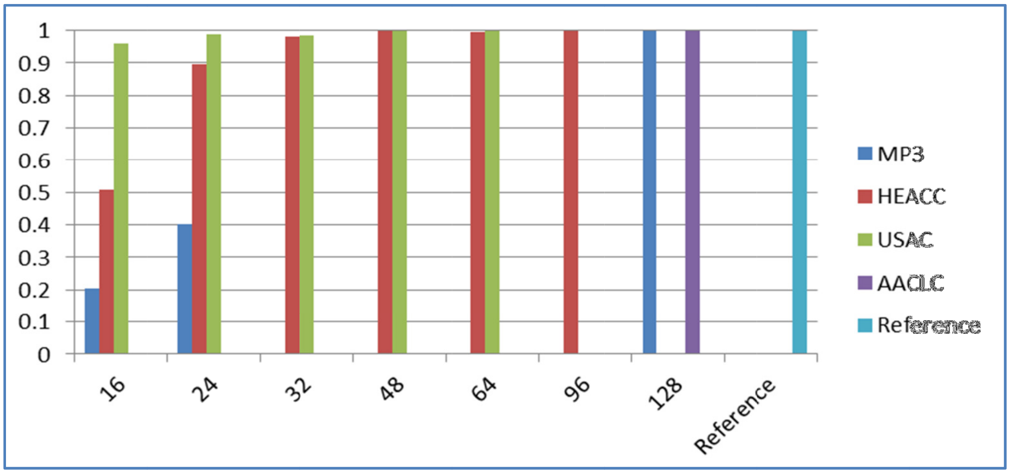
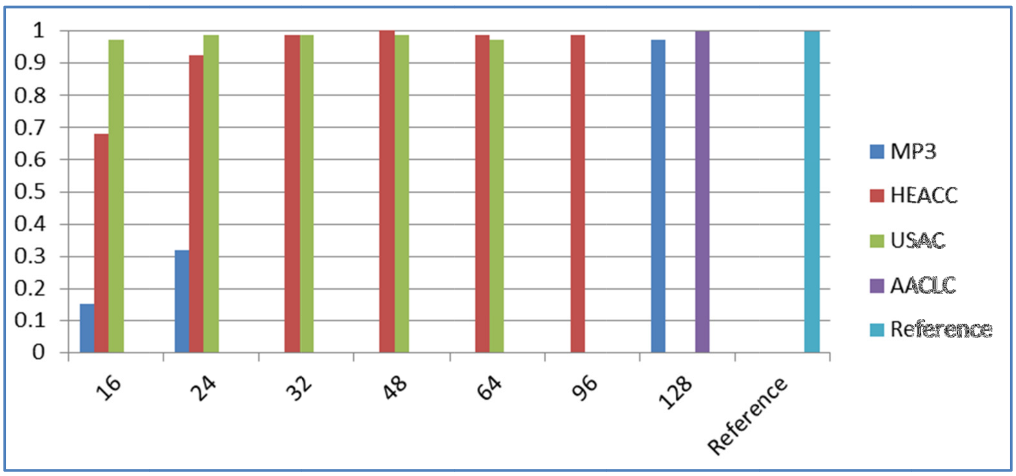


Figure 19 – Leave on - voiceover



LOUDNESS - CONSUMER TEST 2

METHODOLOGY

Forty participants were recruited for the loudness test from Towson University staff and administration via the university's "Daily Digest" announcement forum. See below for demographic details.

Age Group	Female	Male
18-30	5	5
30-40	5	5
40-50	5	5
50-65	5	5

Participants listened to 128 speech and music audio samples, divided into a total of four sessions. In the first two sessions, participants listened to samples via speakers, and then repeated the two sessions listening via ear buds. Speakers and ear buds were counterbalanced across participants so that 20 people used speakers first and 20 people used ear buds first. Participants were encouraged to adjust the volume to a comfortable setting before beginning, after which the volume remained constant.

In the first segment of session 1, participants listened to a single audio sample approximately 15 – 25 seconds in length. Participants were told that some samples would contain a loudness change and some samples would not. They were instructed to press the space bar as soon as they detected a change in loudness, if at all.

In the second segment of session 1, participants listened to two samples, back-to-back, and were instructed to listen for a loudness change between samples. Listeners were presented with different combinations of audio, including speech to speech, speech to music, music to speech and music to music. Samples were approximately 15-18 seconds in length, so that listening to a sample pair took approximately 30-35 seconds. The participants did not have to identify when the change was detected in segment two, as the change occurred when the second sample was played. After each sample or sample-pair participants answered 4 questions:

1) Could you detect a loudness change on a scale of 1-10 (1 – did not detect, 10 – easily detectable);
2) On a scale of 1-10 how annoying was the loudness change (1 – not annoying, 10 – very annoying);
3) If this change occurred while listening to the radio would you do nothing, adjust the volume or turn off the radio.
4) If this change occurred on a regular basis what would you do: nothing, adjust the volume or turn off the radio.

Session three was a replication of session one, and session four was a replication of session two, with the participant listening via speakers or ear buds, whichever they did not use in the first two sessions.

SAMPLE PREPARATION

Audio selections included music and speech. The selections were chosen based on the evenness of their amplitude, using the Orban Loudness Meter with its current ITU-R BS.1770-2 algorithm. Each candidate selection was measured at a 1 sample per second interval and samples were selected that each had a maximum to minimum loudness variation of less than 4 dB. This ensured that listener's scores would not be significantly biased by one part of the selection, such as the last sounds they hear during the sample. Using the audio gain editor in Cool Edit Pro, the selected samples were adjusted in gain so that the average loudness level of the samples was within 2 dB of each other.

ANALYSIS OF AGE AND GENDER

For the loudness test, A 2 (Gender: male, female) x 4 (Age group: 18-29; 30-39; 40-49; 50-65) Analysis of Variance (ANOVA) was again conducted to explore whether men and women were differentially annoyed by change in amplitude. There was no statistical difference among age groups, and no statistical difference between male and female listeners.

WITHIN-STREAM RESULTS

By pressing a button, people were asked to identify where a change in amplitude occurred within a single audio stream. Correct responses were counted only if listeners pressed their buttons within 6 seconds of the amplitude change. Figure 20 shows the percentage of correct and incorrect responses as the dB change increased from 1dB to 9dB. Notice that at 6dB a majority of listeners were able to correctly identify a change in amplitude, but beneath that level, 80% of listeners could not correctly identify a change. People also demonstrated the "false-positive" response, saying that there were changes in samples that were held "constant".

Figure 20 - Percentage of people identifying change in amplitude

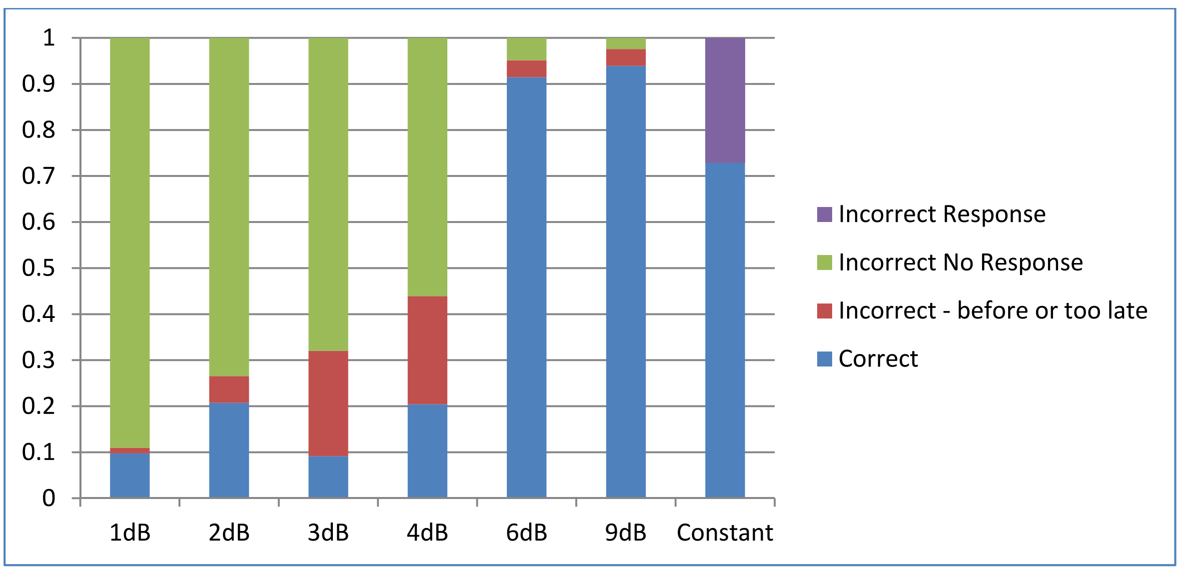


Figure 21 displays participants' ratings of how annoying the change was, on a 1 (not annoying) to 10 (extremely annoying) scale. Notice that participants did not think any of the amplitude changes were very annoying or extremely annoying, but that at 9dB participants claimed that the change was becoming annoying.

Figure 21 - How annoying change was

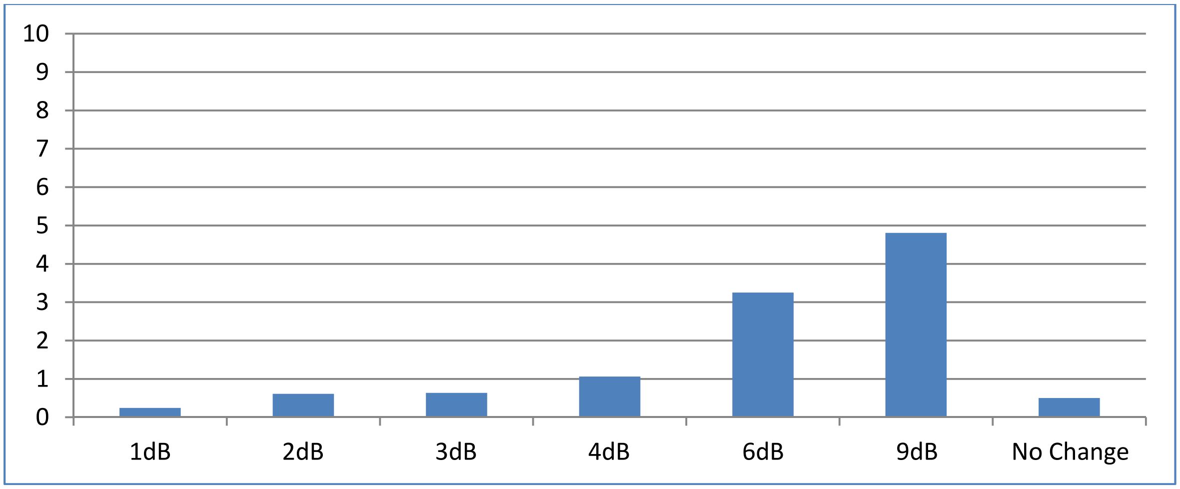


Figure 22 and Figure 23 show how listeners would respond to the radio after a change occurred within a stream. Figure 22 shows that if listeners believed the change were atypical, few would turn the volume off, and instead opt to adjust the volume knob. However, as is seen in Figure 23, if they believed the change would occur frequently, more reported they would turn the radio off.

Figure 22 - Listeners' behavior after a change occurred IF the change was rare

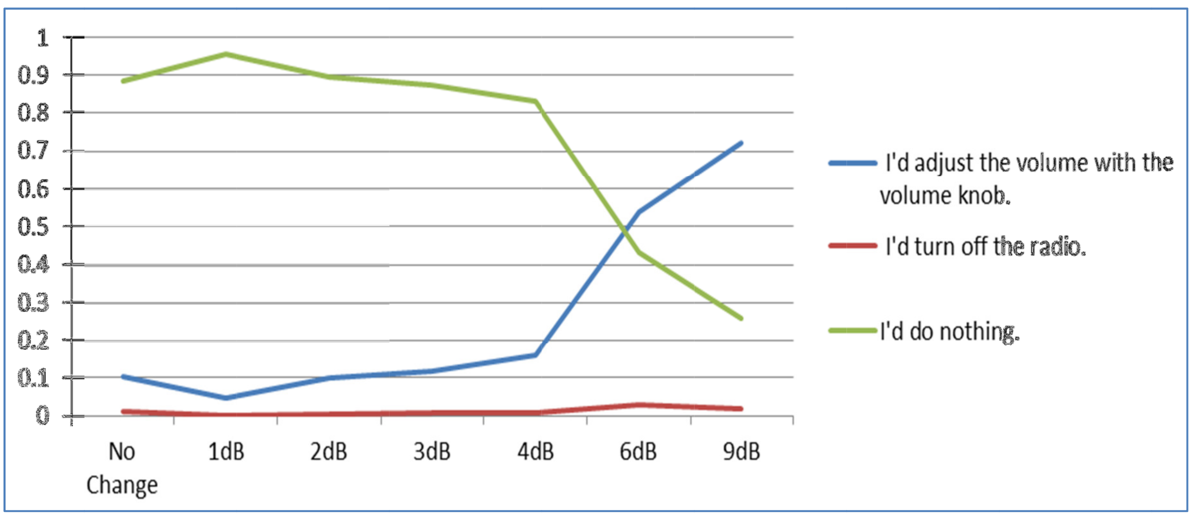
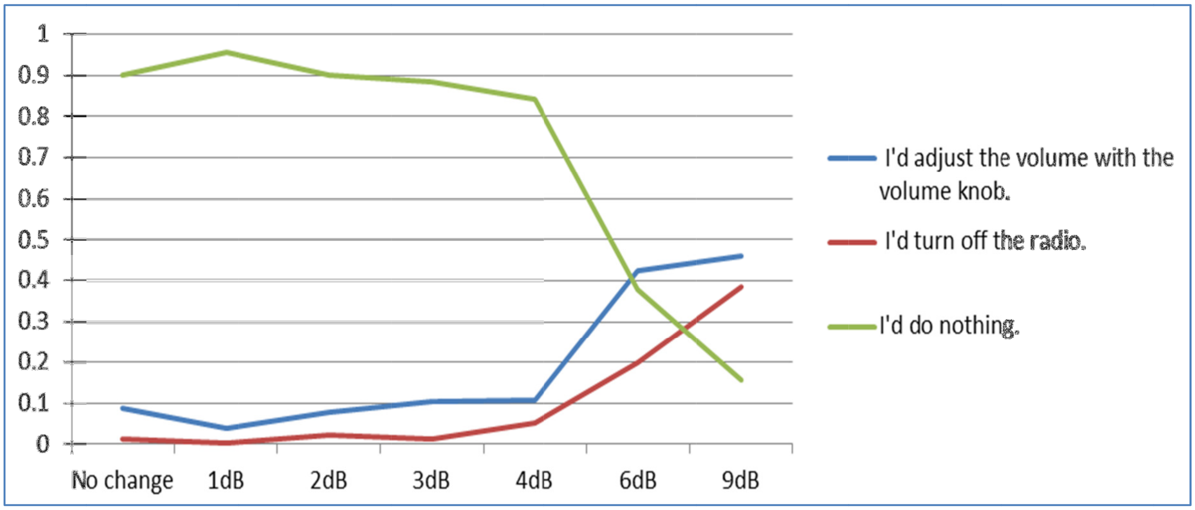


Figure 23 - Listeners' behavior after a change occurred IF the change was frequent



BETWEEN-STREAM RESULTS

Listeners were asked several questions after listening to two audio streams. First, they were asked how prominent the change was. Figure 24 shows listeners ratings when listening on ear buds or loudspeakers. Notice that when the changes were small, participants claimed that they were easier to hear on ear buds. However, when the change was large, both ear bud listening and loudspeaker listening was equivalent. Similarly, as seen in Figure 25, listeners were slightly more annoyed at lower loudness changes when they listened over ear buds, but this annoyance evened out as the change became larger.

Figure 24 - How prominent change was when listening on ear buds or loudspeakers

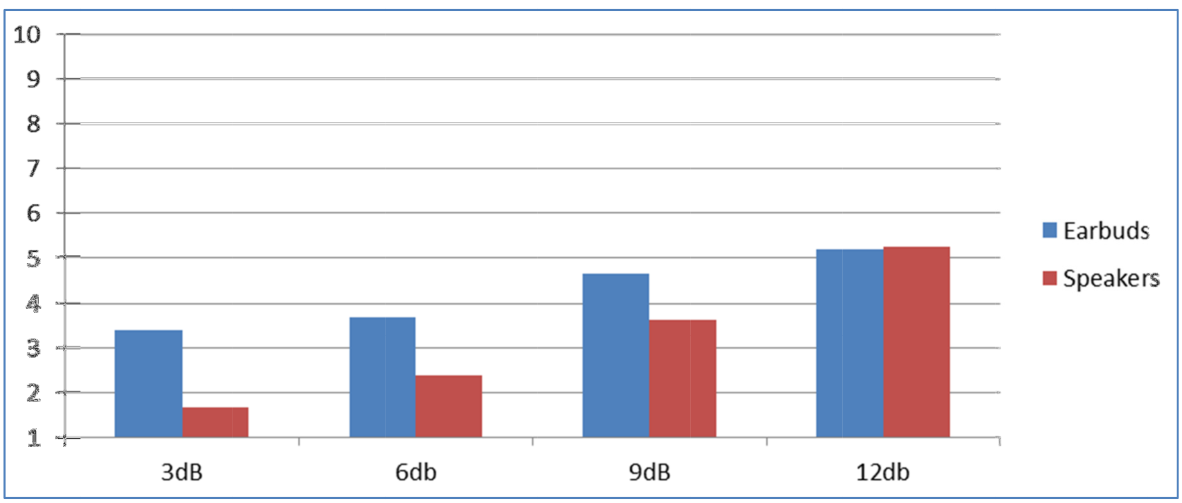
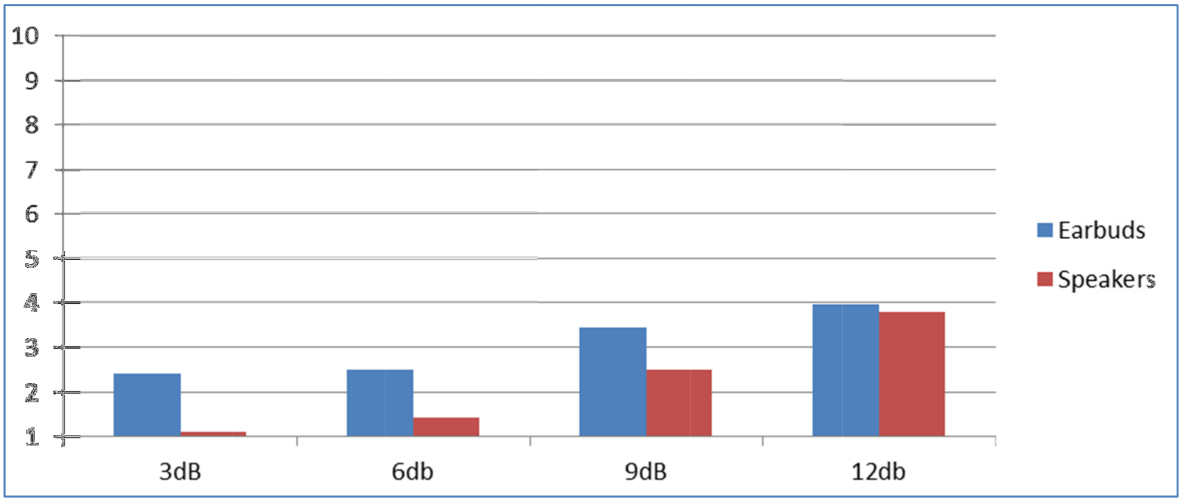


Figure 25 - How annoying change was when listening on ear buds or loudspeakers



With regard to the direction of the change – loud to soft or soft to loud - Figure 26 and Figure 27 show that as the change increased more listeners reported hearing the change and being annoyed by the change if the audio went from softer to louder.

Figure 26 - How prominent change was

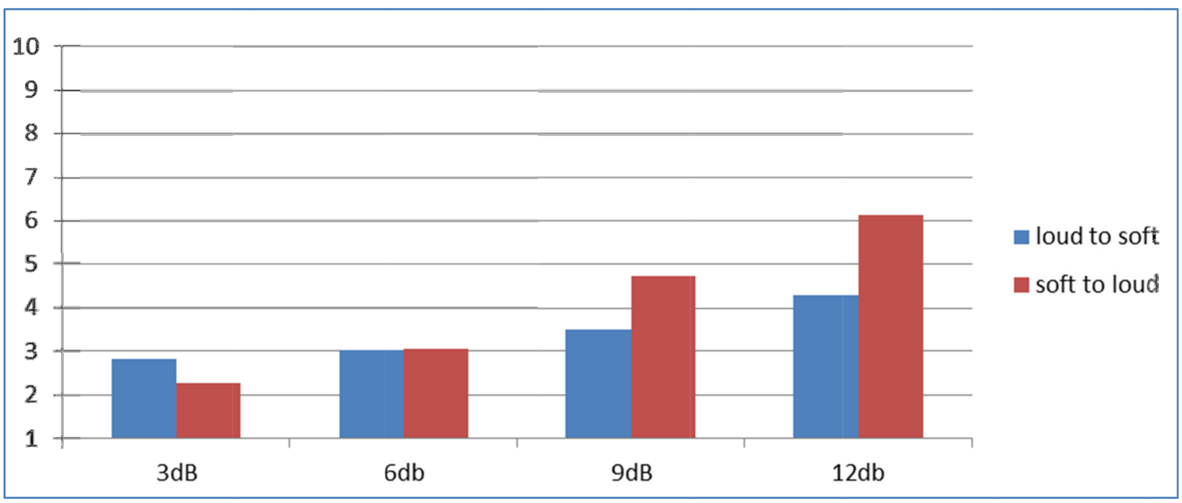
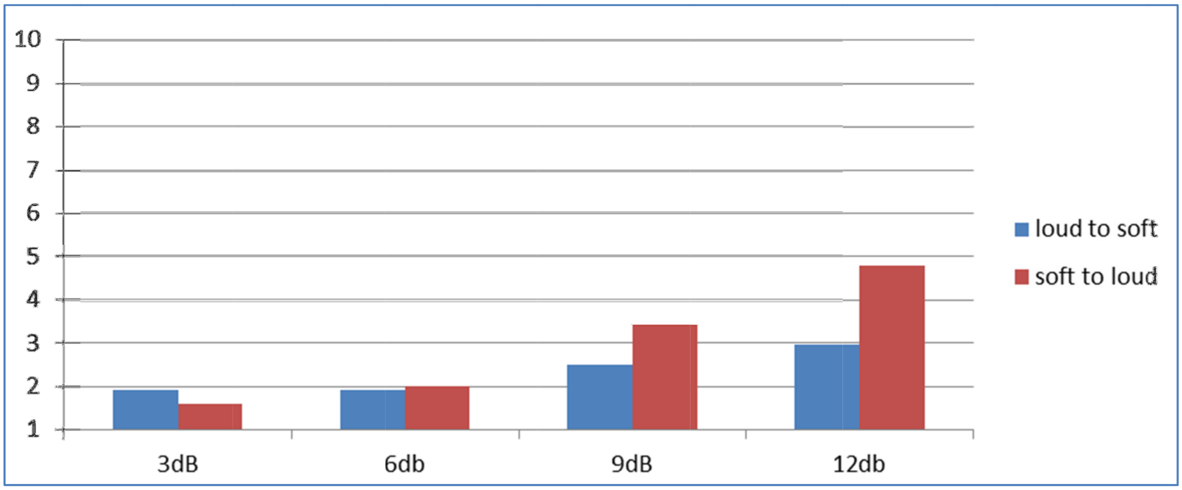


Figure 27 - How annoying change was



With regard to participants behavior, as with the first segment where change occurred within the audio selection, when change occurred between two audio selections participants were more likely to turn off the radio when they felt this change would occur frequently. As is evident in Figure 29, at 6dB change people were beginning to take action, by adjusting the volume or turning the radio off.

Figure 28 - Behavior if radio loudness change occurred once

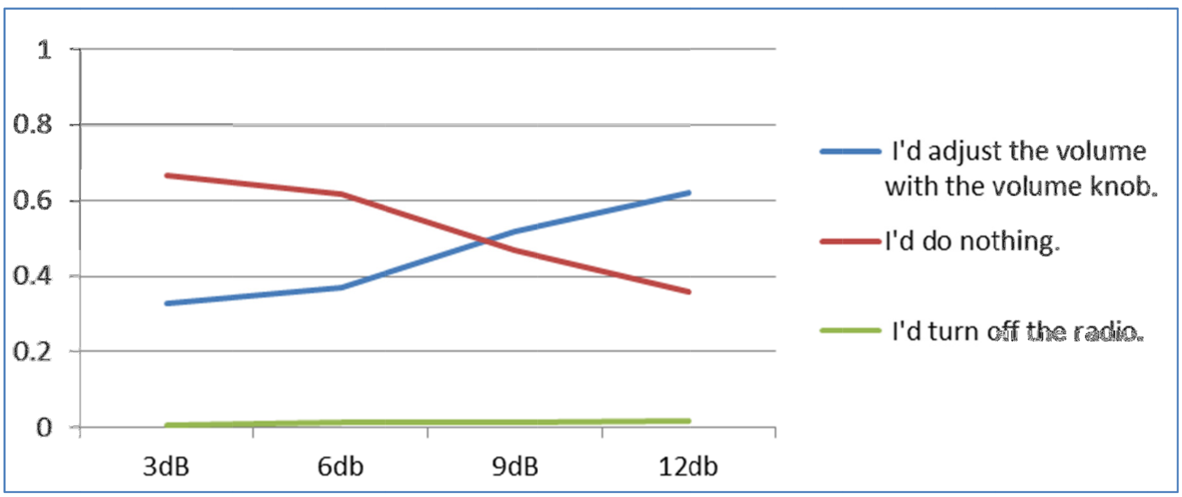
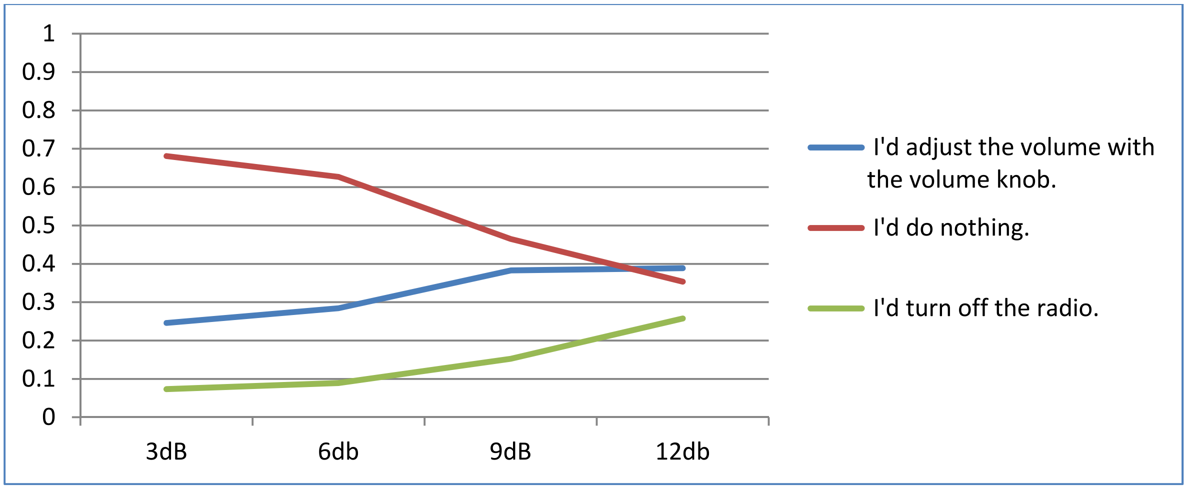


Figure 29 - Behavior if radio loudness change occurred regularly



Finally, when we parsed the data into specific kinds of audio, participants were significantly more annoyed when change occurred between speech samples, as can be seen in Figure 30. It is likely that people are significantly more sensitive to changes in amplitude between speech samples because they expect speech to be more even, and are more familiar with change in amplitude with music, as demonstrated in Figure 31and Figure 32.

Figure 30 - Annoyance between audio streams - speech to speech

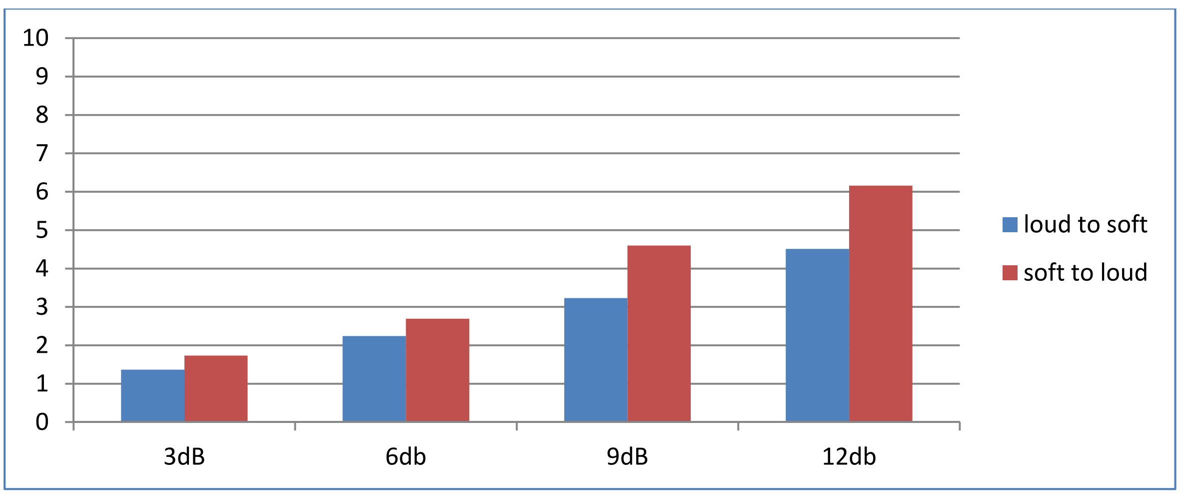


Figure 31 - Annoyance between audio streams - music to music

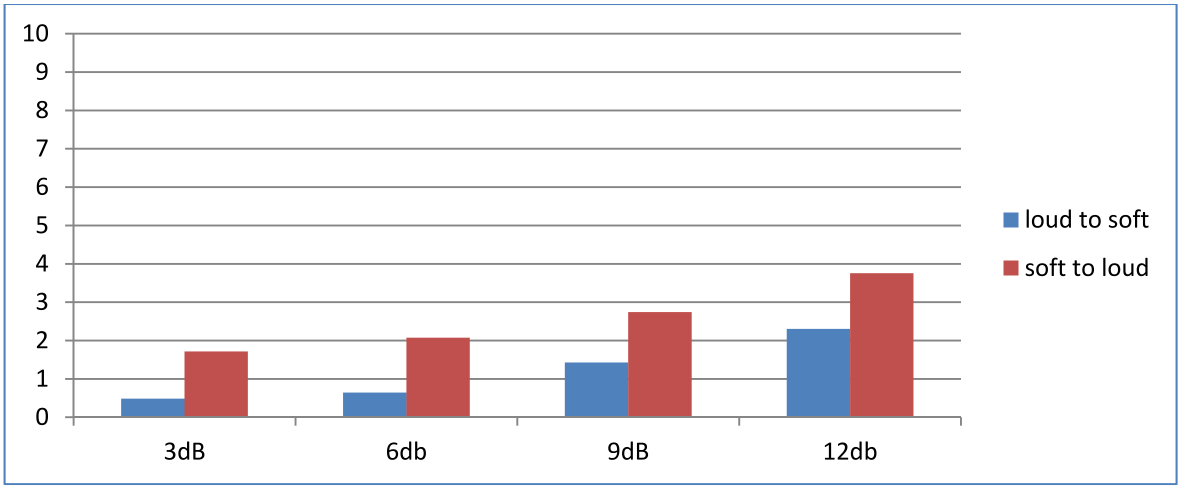
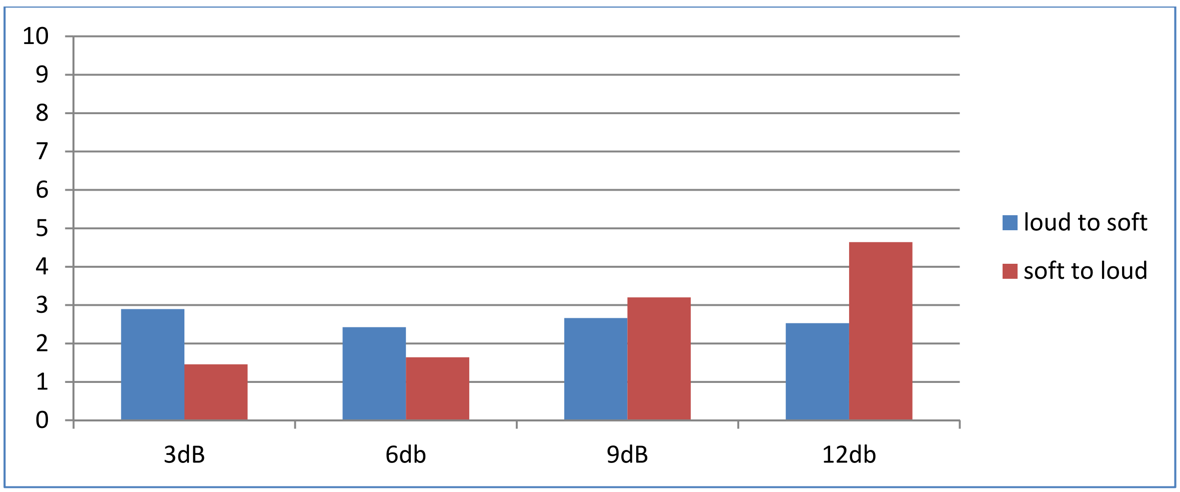


Figure 32 - Annoyance between audio streams - speech/music and music/speech



TECHNICAL RECOMMENDATIONS

The section addresses key factors that affect the quality experienced by listeners, summarized below:[4]

- Absence of digital compression artifacts in the audio (by choosing the right bit rate in association with a given codec);
- Optimum dynamic range to hear the program material, as determined by the listener's needs (controlled, for clarity in a noisy environment, or "open" when the listener desires it and the ambient noise level is lower); and
- Reliability of the stream audio (of less importance for personal computers connected to broadband networks, but a significant concern for streaming over mobile networks).

CODEC SELECTION AND BIT RATE

The two test results discussed earlier in this report provides direct guidance on the selection of a codec and the bit rate. The first study, by experienced listeners, examined a variety of commonly used codecs. In terms of efficiency (the lowest bit rates required to render high mean opinion scores) for a range of audio material, this study identified the USAC and HE-AAC codecs for first and second place, respectively. USAC and HE-AAC provided similar results for music, but USAC excelled at voice quality at the lowest bit rates. As discussed in the section *Audio Codecs Under Test*, however, USAC is not currently available across all smartphone and PC platforms. The HE-AAC codec is sufficiently available in consumer devices, in our opinion, to receive the nomination.

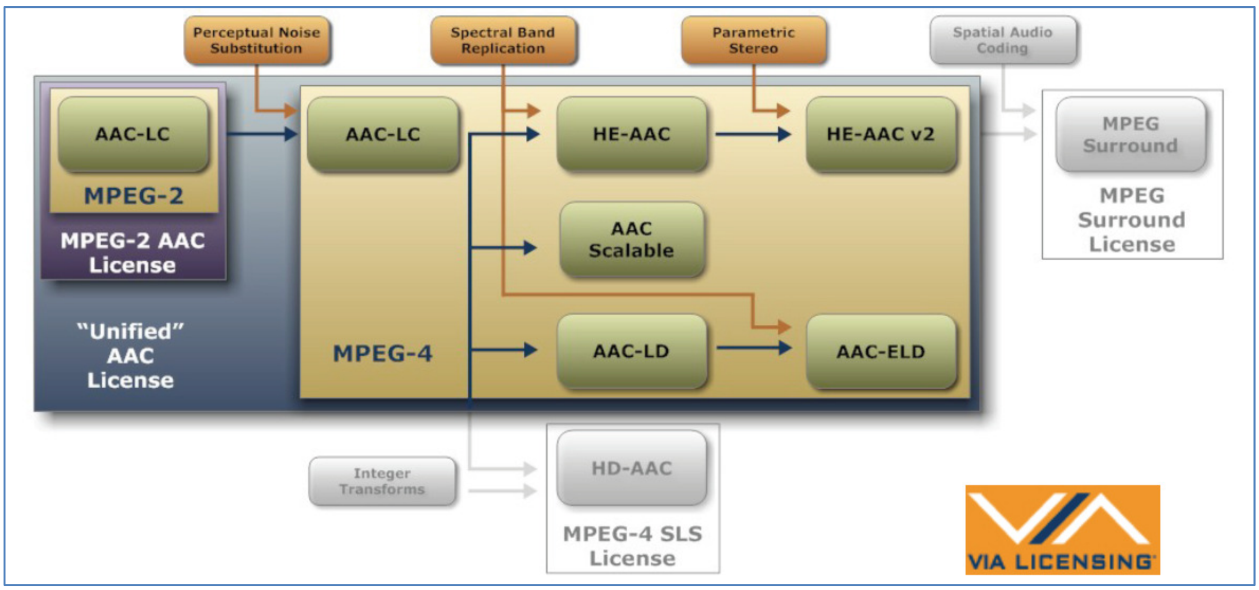


Figure 33 - The Advanced Audio Codec "family"

[4] The quality of the audio itself, that is, to be audibly free from distortion, noise, etc., is under the control of program originators and is beyond the scope of this study

Figure 33 illustrates the codecs under the unified AAC patent license. The codecs are arranged in order of decreasing age, as well as their increasing bit rate efficiency, from left to right.[5] The AAC codecs recommended for NPR's streaming services are contained in the "MPEG-4"box. The AAC-Low Complexity (AAC-LC) codec is appropriate for rates of at least 96 kbps. It spawned the "High Efficiency" (HE-AAC) version by adding Spectral Band Replication. At bit rates below approximately 96 kbps, AAC switches to the HE-AAC mode. This transition is automatic, but the specific bit rate may be modified by sending configuration parameters to the encoder. At rates of 32 kbps and below, Parametric Stereo processing is added, to further improve coding efficiency and extend usable stereo operation to as low as 20 kbps. All three codecs are supported within one software encoder and decoder, and their licensing terms are shared with other forms of AAC, such as AAC-Low Delay (AAC-LD).

CODEC COMPATIBILITY AND LICENSING

The HE-AAC codec is almost universally available in portable devices operated by iOS and Android (at least since version 3.1 "Honeycomb", according to the developer).[6] Android devices before v3.1 may not have HE-AAC on board, however, as these devices are now more than two years old, this percentage is declining as consumers replace their devices with newer models and later versions of Android OS.

Other systems that support HE-AAC include Blackberry, Windows phone®, Symbian® OS and the 3rd Generation Partnership Project (3GPP), which is a collaboration of telecommunications entities. Services such as Pandora®, Netflix®, hulu® and the BBC iPlayer use HE-AAC, as well as global broadcast systems such as Digital Video Broadcasting (DVB), World Digital Multimedia Broadcasting (DMB), Digital Radio Modiale (DRM).[7]

In addition to the technical performance and availability, NPR should consider any costs associated with the codec choice. HE-AAC has no fee, either for NPR to stream or for listeners to use, provided that the codec was supplied by the phone's manufacturer. For those mobile devices that do not include the HE-AAC codec, the AAC codec could be included in a player-software download, but the developer (NPR) would be subject to mandatory license fees: a license fee of $15,000 for five years plus $0.98 per download for the first 500,000 units.[8]

---

[5] These designations are supplied in a figure by the licensing agent, Via Licensing Corporation, which represents a number of patent holders involved in the development of AAC. The designations may vary.

[6] The page at http://developer.android.com/about/versions/android-2.3-highlights.html states (on 10/2012) that "the [Android 2.3] platform also adds support for AAC encoding and AMR wideband encoding (in software)". Decoding would also be needed to playback one's own files, although this is not stated. The supported files and container formats at the page http://developer.android.com/guide/appendix/media-formats.html list "ADTS raw AAC (.aac, decode in Android 3.1+, encode in Android 4.0+". This statement seems inconsistent with the first reference.

[7] *HE-AAC* – The Universal Solution, Fraunhofer IIS presentation to the Audio Engineering Society, October 2010.

[8] Licensing on behalf of the patent holders is administered by Via Licensing Corporation; see http://www.vialicensing.com/licensing/aac-faq.aspx and http://www.vialicensing.com/licensing/aac-fees.aspx.

The MP3 format, while universally available, appears to be an alternative. However, the drawbacks to that codec may include:

- MP3 streaming requires at least 2½ times the bit rate of HE-AAC for similar quality, which increases streaming costs and may accumulate higher data fees for listeners;[9]
- The higher streaming rate needed by MP3 will lower stream reliability, compared to HE-AAC.

A cost-benefit analysis is indicated for the case of HE-AAC codec download versus the cost of an alternative MP3 stream. The following example shows that, depending on the number of listeners who download the NPR app with HE-AAC (we assume the current version has been downloaded at least 100,000 times), the cost to NPR of the codec license could be made up in a matter of time through streaming cost savings by using HE-AAC instead of MP3. The example below illustrates the number of months to break-even for the AAC license if the added streaming cost were 5¢ per user per month.

overall license fee		$15,000		
per download		$ 0.98		
number of downloads:	10,000	50,000	100,000	500,000
cost per download	$ 2.48	$ 1.28	$ 1.13	$ 1.01
difference per user per month, 128 kbps		$ 0.05		
time it would take to get to the download cost for the AAC decoder				
(number of months)	49.6	25.6	22.6	20.2

Another consideration for implementing any advanced codec is computers running older operating systems, such as Windows XP, which lacked AAC. Codec upgrades that include a compatible AAC decoder are widely available for download on the Internet, and it may be possible to use separate public radio stream software to avoid the licensing fee for consumer PC software, which is $0.48 per unit plus $32,000 per year.[8] Compatibility of HE-AAC with older Apple computer products is not likely to be an issue as Apple's iTunes uses the AAC codec exclusively.

[9] The StreamGuys hosting service (www.streamguys.com) levies a $0.05 extra cost per user per month (at 1000 user capacity) for customers who stream at 128 kbps, compared to those who stream at 48 kbps.

STREAMING REQUIREMENTS

While it is undeniable that that higher bit rates permit potentially greater audio quality, there are other conditions besides audio quality to consider, such as the reliability of the stream, which runs counter to the bit rate. The following section explains how wireless mobile networks affect stream reliability and how this impacts the choice of streaming bit rate.

WIRELESS NETWORKS

Wireless mobile networks, such as Verizon and AT&T, are essential to a mobile audio service. These carriers are licensed by the FCC to provide two-way broadband data service to large geographic areas of the U.S. According to Consumer Watchdog (www.consumerwatchdog.com), however, they promise "faster 4G" speeds "without actually making improvements to existing products and services or without disclosing the meaning of 'faster'". Prompted by complaints from consumers about irregular broadband speeds and uneven geographic service, the FCC announced that it will test mobile operators' systems in the near future (see Figure 34). Some industry analysts doubt that much can be done to improve performance, as customer growth and user demand for greater amounts of data will continue to soak up the networks' 4G and LTE speed improvements.

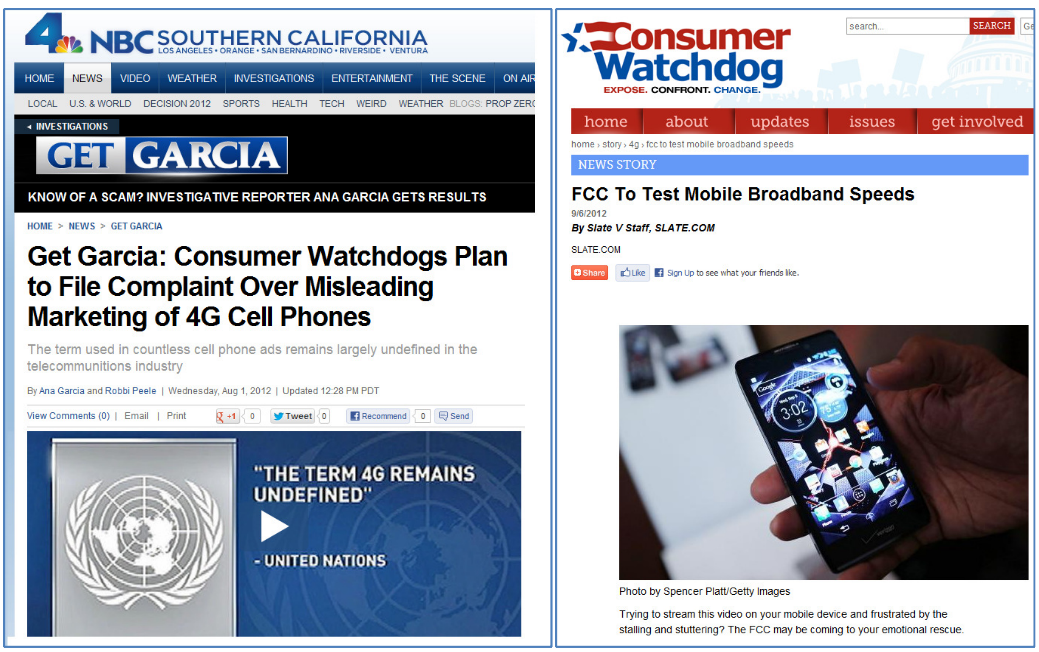


Figure 34 - Examples of news reports about the misleading speed claims of mobile wireless operators; FCC intention to investigate

The bandwidth of wireless networks is not an absolute: it can be described by averages, with statistical variations around a mean. The measurements in Figure 35 further illustrate that bit rates, when grouped into ranges, vary in their occurrence (frequency).[10] The measured rates compare UDP, an Internet transport protocol with no data retransmission, to TCP, a transport protocol that can request retransmission of data packets. UDP is commonly used for streams while TCP is a core protocol used on the Internet for reliable file transfer.

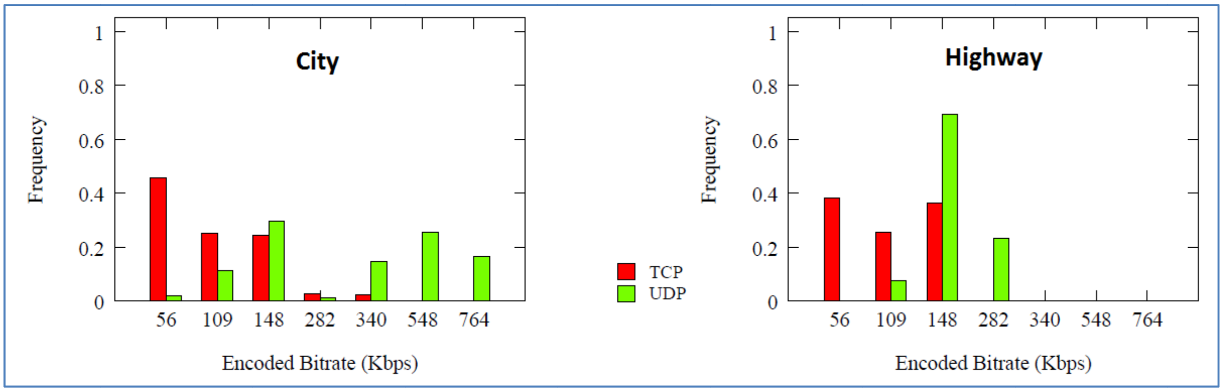


Figure 35 - - Statistical availability of bit rate for TCP and UDP protocols

Reliable protocols, such as TCP, guarantee correct delivery of each bit in the media stream. However, they accomplish this with a system of timeouts and retries, which makes them more complex to implement than UDP.

UDP is it preferred for streaming because it has less latency and processing overhead than TCP, but it is less reliable because packet checking and retransmission requests are not possible. With either protocol, when there is data loss on the network, a media stream stalls while the protocol handlers detect the loss and retransmit the missing data. Clients can minimize this effect by buffering data, but at the expense of longer startup times and latency.

It is apparent that UDP offers higher more occasions for higher downstream bit rates than TCP. The Figure 35 charts show that bit rates in the city have a wider variability than the highway measurements. It is important to note that these measurements are averages over time, and that the data, especially with UDP, could include shorter intervals when no data is received.

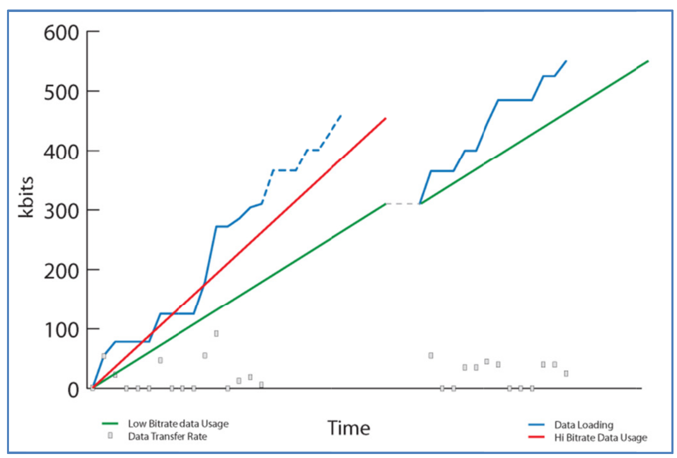


Figure 36 shows a simplificatiion of how audio data packets can flow over an Internet network (wireless or wired) to a UDP device, as the small blocks along the base line. The accumulated data over time is shown with the blue line. If the audio playout rate, shown by the red line,

[10] The Effects of Mobility on Streaming Media in CDMA2000 1xEV-DO Network, Phillipa Sessini, et. al., Dept. of Computer Science, University of Calgary, *Proc. SPIE*, Jan. 2007.

exceeds the received data, dropouts in the audio will occur. If a lower audio playout rate, shown in green, is less than the rate at which data accumulates in the buffer, uninterrupted audio results. These scenarios do not apply to file-download systems, such as Pandora, which are effectively file transfer systems with a large data buffer: the data transfer can even stop, briefly, as long as the data bursts keep ahead of the file playout rate. (It is believed that Pandora and similar file-playout systems may also shift to a lower playback bit rate in anticipation of a data short fall, which helps avoid audio dropouts. Because podcasts are file transfers, their playback after download is complete; hence, high bursty bit rates are possible without effect to the playback.

Prior to this project we constructed an in-vehicle system to measure the reliability of different streams. To do this we connected the audio output from a smartphone to an audio sensor and computer that could record time and location along with audio availability. The phone was placed in the clear, in a holder at the top of the dash, and a GPS antenna was placed on the roof. Our primary interest was to determine if stream rate would affect reliability.

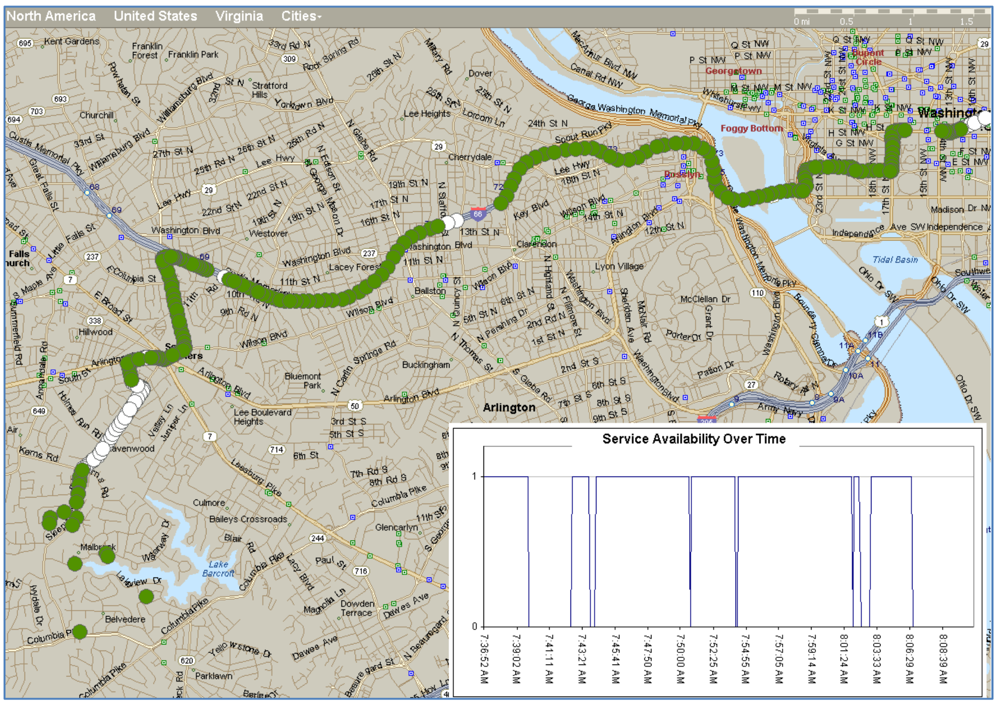


Figure 37 - Typical route, between NPR and home for one of the authors.

The map in Figure 37 shows a trip to NPR headquarters, in the upper right, from the Virginia suburbs as a series of dots. The green dots indicate stream audio and the white dots show dropouts, using an Apple iPhone 3GS on the AT&T network.[11] The inset chart shows the audio availability over time for the same trip. It is apparent that this stream (at 128 kbps from NRK Klassik, in Norway) had several dropouts, with two lasting about two minutes each. Our testing found that lower bit rate streams were consistently more reliable than higher streams.

---

[11] The iPhone 3GS introduced support for 7.2 Mbps HSDPA (High-Speed Downlink Packet Access) in the Universal Mobile Telecommunications System (UMTS), a third generation mobile cellular system for networks based on the GSM standard. This capability would appear to far exceed the 0.128 Mbps (128 kbps) stream rate in the example.

The stream logging system was expanded later to support two smartphones. In that way we could by simultaneously measure two different streams in one vehicle during the same trip, cancelling out the variations that may occur from trip to trip. Figure 38 shows two member stations with stream rates in this test of 32 kbps (upper, in red) and 128 kbps (lower, in blue), both received on iPhone 4's on the AT&T network. The higher bit rate stream experienced more dropouts, including one 17-second drop at approximately 573 seconds from the start.

Dropout behavior with higher bit rate streams was frequently less reliable, but not always consistent. As Figure 38 shows, the dropouts are not correlated in time, suggesting that in populous areas where the signal is consistently high, the dropouts are caused by network capacity limitations.

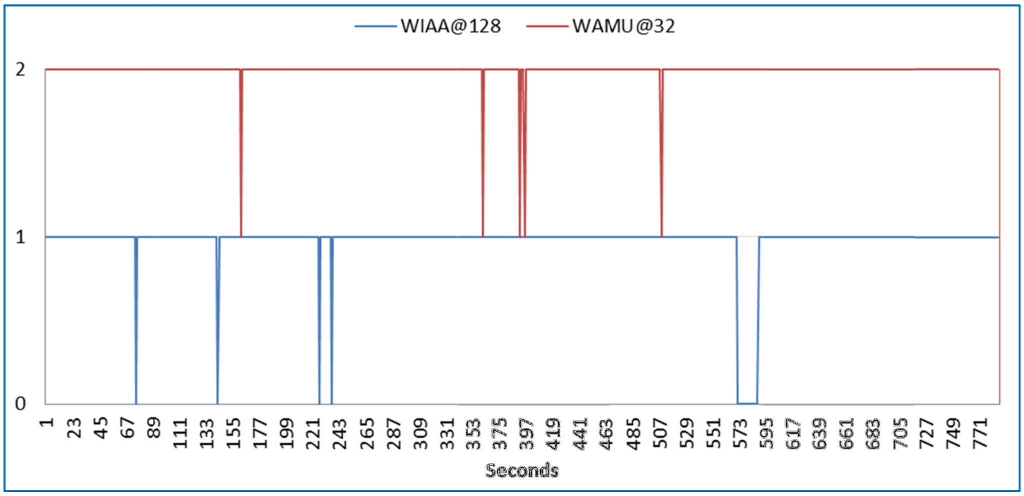


Figure 38 - Comparison of dropouts on simultaneous streams at 128 and 32 kbps

Our tests found significantly higher dropout during peak morning and evening travel time than during off-peak times, which reinforces our theory that network loading, not signal, was reducing stream reliability. This may be due to travel from one wireless cell into another cell that is at temporarily at capacity, where no other cell is available for handoff. We also found cases where handoff appeared to be delayed, such as the section of I-66 leaving Washington and entering Arlington, where we theorize that handoff between Washington and Virginia networks is occurring that causes regular and predictable dropout. The dropouts are usually more prolonged with high bit rate streams in this section of roadway.

CODEC AND BIT RATES

From the testing discussed above, we conclude that lower bit rates offer noticeably higher resilience against momentary network capacity issues and poor cell handoff. The effect of dropouts on listener satisfaction was not measured in this study, but we presume that listeners' annoyance with dropouts rises quickly with frequency or duration, and could lead to "tune outs". **Based on our research, we recommend keeping the bit rate of streams as low as possible, consistent with high audio quality. For the HE-AAC codec, we believe this balance is achieved at a constant stream rate of 48 kbps.** (Variable bit rate is not suited to wireless streaming as the instantaneous bit rate may exceed the data channel capacity, resulting in increased dropouts.)

LOUDNESS CONSISTENCY

Next to digital audio quality, the uniformity of perceptual loudness, both from stream-to-stream and from program-to-program within streams, is one of the most important factors to listener enjoyment. In television, for example, the matching of commercial announcements to programming and control of dynamic range have required large investments in technology research and development to tackle this problem to consumer's satisfaction. Streaming services for public radio deserves special consideration because of the wide variety of environments in which listeners may hear streams and because they can easily switch between streams of varying program types.

As explained below, the current consistency of loudness between public radio streams is poor. This appears to be due to multiple factors, including a lack of standards for digital audio transmission level, inattention to audio levels that inadvertently get out of adjustment, and great variety in audio processing devices and the way they are operated. At the same time, audio streams on the Internet unrelated to public radio operate across an enormous range of audio levels, which presents issues for listeners who may switch to and from public radio streams.

Listeners may hear public radio streams in a variety of environments, from a noisy subway car to a quiet living room. Those in noisier locations need tight control of dynamic range to hear every word of a talk show or newscast, and every note of a symphony. However, the degree of loudness management necessary for noisy listening conditions is entirely different than in a quiet room. Consequently, we present a solution that puts the active loudness management in the player, where the listener can choose the dynamic range that's suitable for their environment and taste.

CURRENT EXPERIENCE

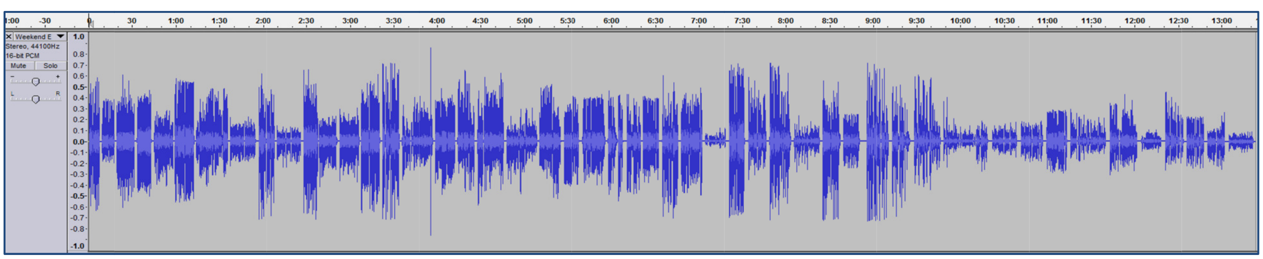


Figure 39 - Audio samples of 49 public radio stations carrying NPR's *Weekend Edition*, Saturday; variation in level exceeds 21dB

Figure 39 is a screen shot of the program levels from 49 public radio station streams, all carrying the same program: *Weekend Edition, Saturday*, on February 21, 2012. Each sample is more than 20 seconds, which affords time to determine each stream's level, which ranges by more than 21 dB. <u>Listeners who switch between some of these streams would hear changes in loudness large enough, according to our research, to be annoying or even cause listener turn-off.</u> This clearly this is challenge for public radio's success with Internet audio streams.

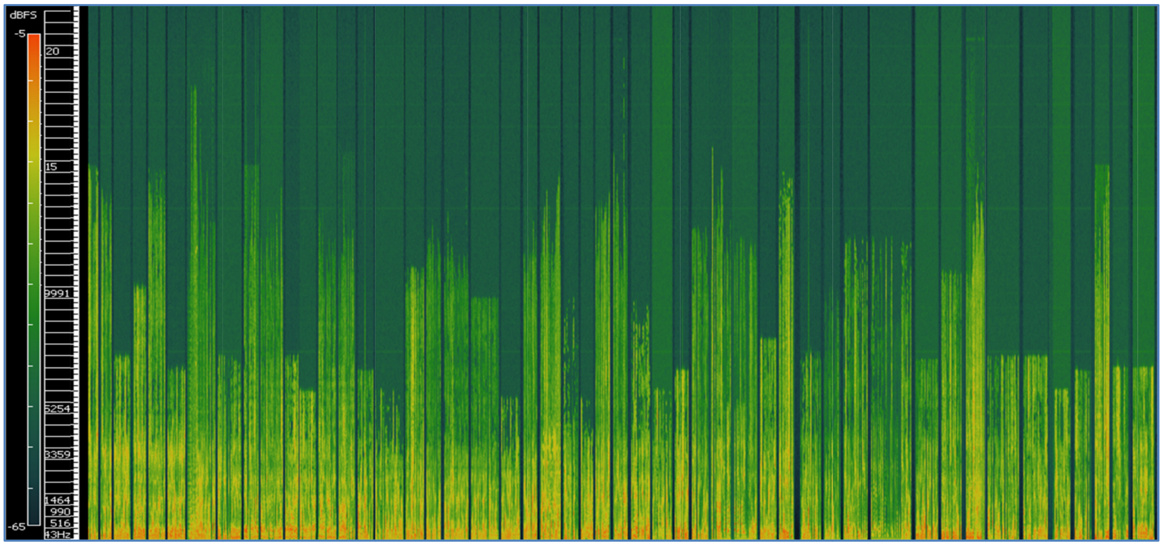


Figure 40 – Audio spectrogram of 49 public radio streams

Figure 40 shows a frequency spectrogram of the same 49 public radio streams carrying WE Sat. The heights of the vertical bars show the highest frequency transmitted, from 43 Hz at the bottom, to 20 kHz near the top, with 9991 Hz near the center. Colors indicate the amplitude, from red (highest amplitude) to dark green (no energy). Several streams contain substantial energy up to 15 kHz, while several others roll off just above 5 kHz. These variations are due to different bit rates for the streams, as well different codecs. <u>Listeners switching from stream to stream will hear large variations in the quality and naturalness of speech and music.</u>

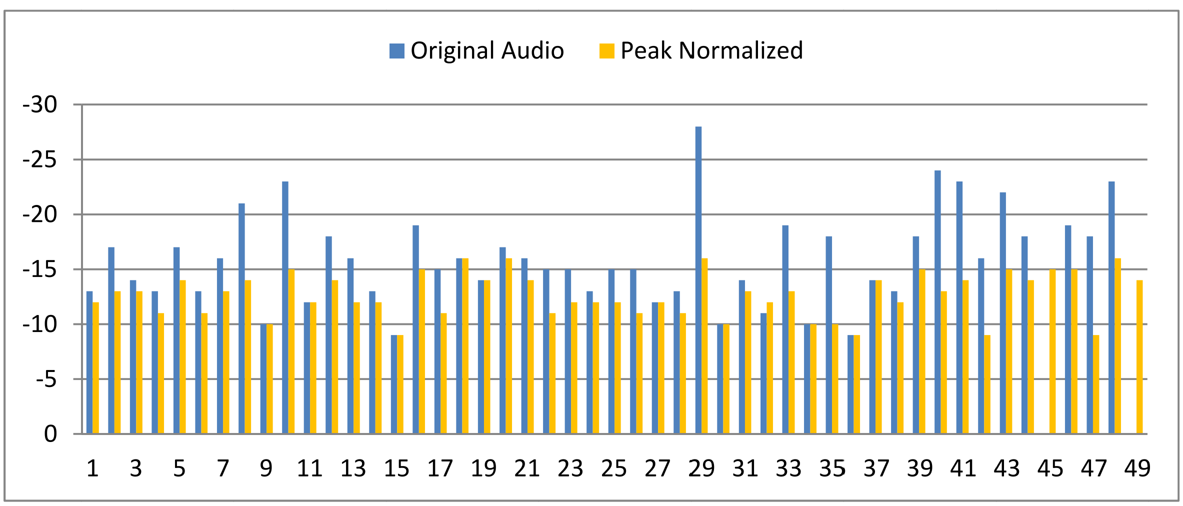


Figure 41 - Loudness of same audio samples as above (CBS Loudness Meter) in blue; orange bars show level variations after each sample was normalized according to its electrical peak values.

Figure 41 is a chart of loudness (measured by the CBS Loudness Meter) of the 49 public radio streams before and after "peak normalization". This technique adjusts the audio gain of each stream sample to produce the same electrical peak level – a common practice, especially in the editing of digital audio, as the entire file can be examined. However, this technique is also similar to most audio processors (compressor-limiters), which adjust their gain dynamically to produce a <u>constant electrical peak level</u>.

The difference in height of the peak normalized (orange) bars to the original (blue) bars shows that despite peak normalization, the loudness varies by up to 8 dB across the streaming samples. This demonstrates that even when audio is "leveled" according to peaks, there can still be variations in loudness that could be unnatural, or even annoying to listeners. **We recommend that audio production implement an EBU 128/ITU-R BS.1770-2 loudness measure, which imitates the way humans perceive loudness, rather than relying on electrical peak level alone.**[12] This is discussed further in the next section,

Even if the all the audio processors were the same and adjusted alike, they only provide uniform peaks – not necessarily uniform impression of loudness. The television industry has developed automatic loudness controllers for program audio, based around the International Telecommunications Union standard ITU-R BS.1770-2, "*Algorithms to measure audio programme loudness and true--peak audio level*". This standard defines a loudness measurement procedure that generally follows human hearing. It also has been used to build a system to regulate loudness for broadcast and streaming applications. Following our report on tests of the tolerance of listeners to loudness changes, we present recommendations that build on our results and current loudness measurement.

LOUDNESS CONTROL AND MANAGEMENT

PROGRAM LEVEL MEASUREMENT

To ensure live-production streams are encoded with uniform program levels, the following procedures are recommended, based on the scales in Figure 42:[13]

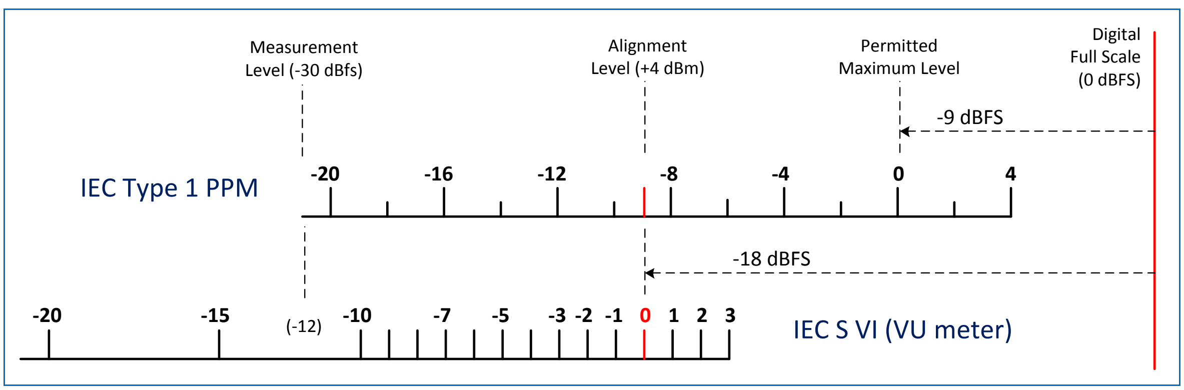


Figure 42 - Alignment and maximum program levels for common meters based on ITU and EBU recommendations

---

[12] Loudness normalization and permitted maximum level of audio signals, EBU Recommendation R128, Geneva 2011, and Practical guidelines for distribution systems in accordance with EBU R 128, EBU Tech 3344, Geneva, October 2011.

[13] This generally follows ITU-R BS.1726, *Signal level of digital audio accompanying television in international programme exchange*, and ITU-R BS.645-2, *Test signals and metering to be used on international sound programme connections*.

- For systems using an IEC Type 1 peak program meter for production, the PPM's Permitted Maximum Level (typically 0 dB on its scale) with a test tone is set 9 dB below the encoder's Digital Full Scale (0 dBFS, shown as the red vertical line to the right);
    - Maximum program peaks will occasionally reach the 0 dB scale level, with 9 dB for unexpected increases in program level;
    - Lineup tone at 18 dB below Digital Full Scale would typically indicate at -9 dB on the meter scale;
    - Measurement tones at -30 dBFS, for example for frequency response testing, would indicate 21 dB below Maximum Permitted Level;
- For systems utilizing a ANSI VU Meter (IEC Volume Indicator), a tone a 0 VU indication is set 18 dB below Digital Full Scale;
    - Maximum program peaks will occasionally reach the 0 VU scale level, with-18 dB of headroom for peak overshoot and unexpected increases in program level;
    - Measurement tones at -30 dBFS, for example for frequency response testing, would indicate 21 dB below Maximum Permitted Level.
- To monitor the level of stream audio we recommend measurement with BOTH an IEC 60268 10 Type 1 meter for quasi-peak electrical level, and an ITU R BS.1770 2 loudness meter. The first meter ensures that the transmission waveform is not clipped and the second meter determines the instantaneous loudness.

LOUDNESS VALUES SHOULD BE THE TARGET, NOT PEAK LEVELS

Error! Reference source not found. shows a comparison between loudness-normalized audio (normalized within a series of 20 to 30 second subsections) and the simultaneous electrical peak level. The loudness ranges around the nominal -23 LKFS value, with a variance of a two or three dB. At one point of maximum loudness (around 400 seconds from the start), the electrical level (shown by the double arrow-heads) is 9 dB

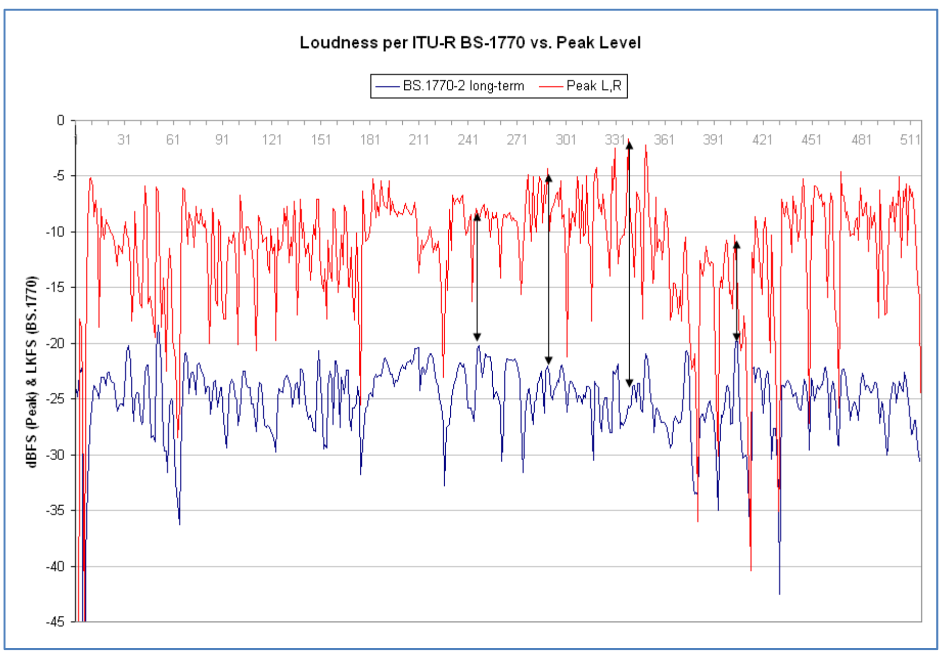
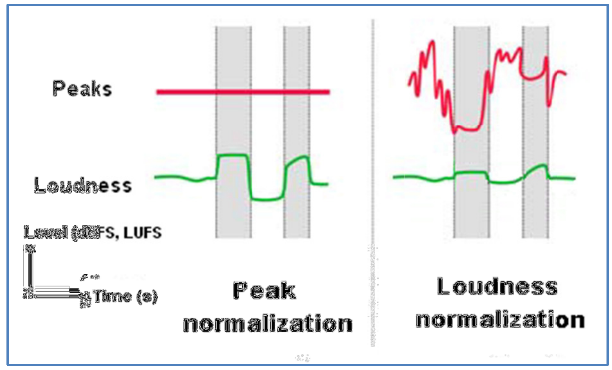


Figure 43 - - Graph of loudness-normalized audio and peak level

above the loudness value. At about 340 seconds from the start, the maximum electrical peak is more than 20 dB above the loudness at the same moment. If we had normalized according to the electrical peaks, it is clear that loudness could be thrown off by many dB. Peak-responding compressors and limiters do that by design. As discussed earlier, the streaming system has headroom similar to the equipment used to produce programming: peak-controlling devices that affect the natural dynamics in program material are not required if the transmission level plan of Figure 42 is observed.

For both program production and stream transmission, loudness of the program should be the primary measurement – peak levels should only be observed to avoid clipping. The left side of Figure 44 illustrates programming that is produced with peak metering, post-processed with peak liming and compression, or peak-normalized; the red line shows uniform peak level from program to program. However, the loudness, shown by the green line, varies up and down. In the right panel, the loudness of each program is evaluated and its gain is increased or decreased to produce a nearly-uniform result. The electrical peaks have been shifted by the gain changes, which produces non-uniform levels. If sufficient headroom is built into the system, the upward displacement in peaks will not be a problem.



vel

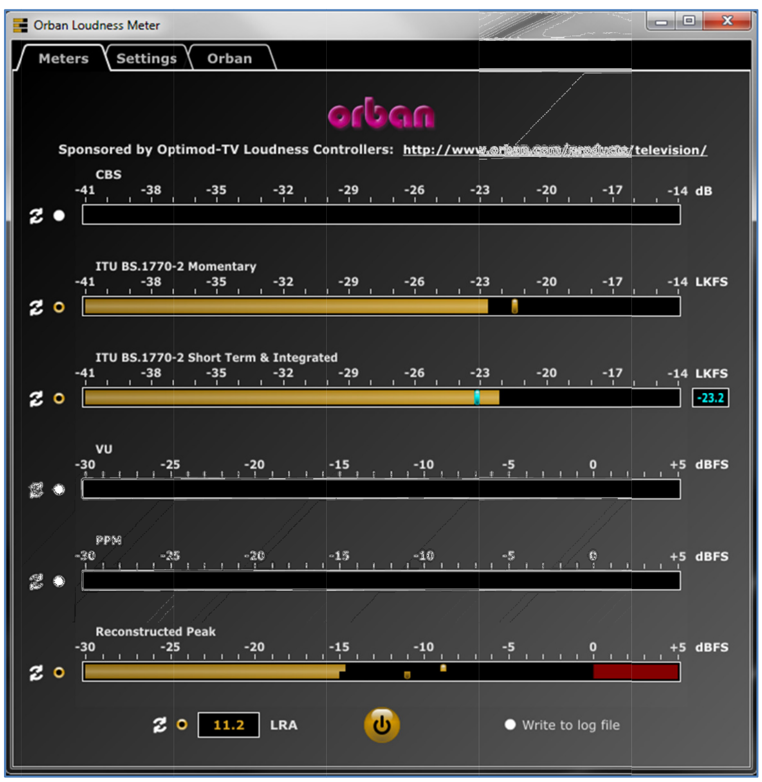


Figure 45 - Orban Loudness Meter v 2.0.8, showing ITU BS.110-2 and Reconstructed Peak measurement bars

For engineering work, an excellent free software meter that provides the specified and other metering standards is the Orban Loudness Meter for Windows computers (http://orban.com/meter/). This software provides real-time display as well as the ability to record the measurements to a CSV file for later analysis. It includes a "Reconstructed Peak" meter that indicates the peak value following D/A conversion, which can cause clipping as they may be higher than the digital samples before conversion. Caution is advised in using this meter with the Windows XP operating system, as its calibration is affected by a combination of controls in Windows' audio mixer.[14] The Orban meter is excellent for technical monitoring, but its window is large and cannot be resized or simplified. We recommend the use of meters designed for audio workstations production editors that provide a simpler loudness indicator, such as the examples in Figure 46 and Figure 47.

Figure 46 - LevelViewS provides simple EBU/ITU loudness metering in live and integrated (long-term) forms.

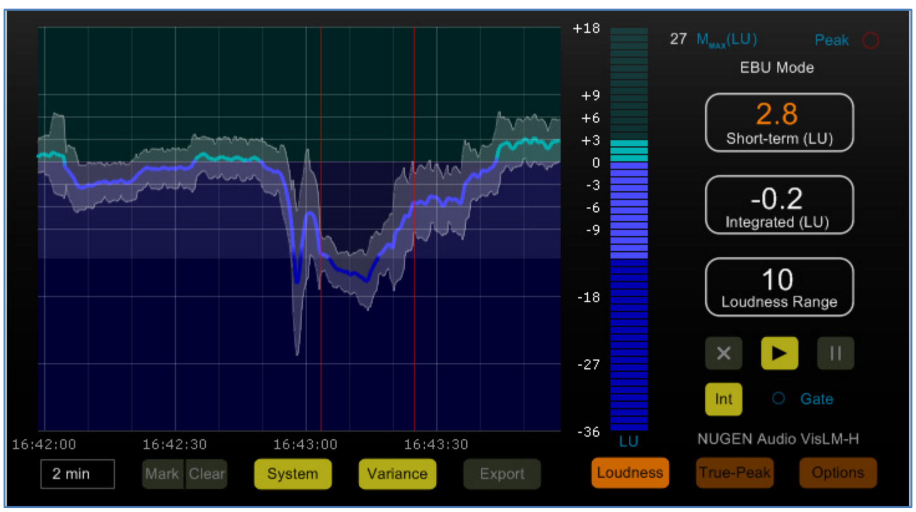


Figure 47 - The NUGEN Audio VisLM-H provides both real-time and historical charting of audio levels.

---

[14] For computers running the Windows XP operating system, it is very important to calibrate your meter device through the Windows Wave I/O or Stereo Mix, to ensure accuracy and to prevent the sound device driver from clipping. For Windows Vista and 7/8, the meter uses Windows Audio Session API (WASAPI) Loopback, which allows the meter to monitor play out from any Windows Audio player application and eliminates the Wave I/O or Stereo Mix requirements. Level calibration is unnecessary because the WASAPI Loopback is internally unity-gain.

LOUDNESS MONITORING AND NORMALIZATION OF STREAMS

Stream validation, the process of monitoring audio streams for quality, can easily incorporate audio level monitoring to ensure that audio is transmitted with acceptable ranges. There are several firms that do that commercially, especially for audio that accompanies video. Fortunately, the need to manage loud commercials while offering pleasing amounts of dynamic range had led to considerable development in video production and transmission.[15] There are tools available for real time monitoring and reporting of audio stream levels, such as the product illustrated in Figure 51. This type of software could be used to spot audio level problems, while the stream validations checks digital quality, metadata, and other aspects of each stream. W**e recommend that automated loudness monitoring be included in the stream validation system. As discussed below, the monitored levels provides information to adjust and normalize the audio transmission level of any streams that To ensure a pleasant experience for listeners, the audio level of all streams should achieve a long-term loudness target of -23 LKFS.**

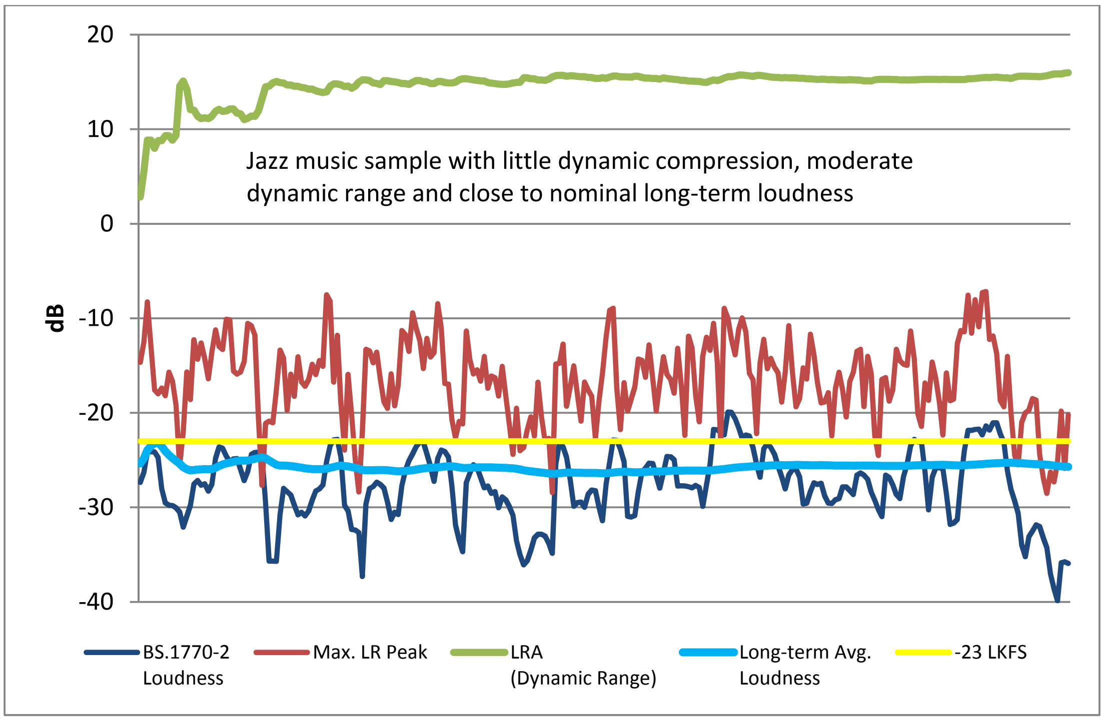


Figure 48 - Several minutes of jazz music; BS.1770-2 loudness, peak and long-term loudness are relative to 0 dBFS

---

[15] In the U.S., in response to the CALM (Commercial Advertisement Loudness Mitigation) Act, the Advanced Television Systems Committee established a study group to develop a "recommended practice"A/85 that includes measurement, metadata insertion into audio streams and loudness management at the consumer's television. The Europeans formed the EBU Group "P/LOUD" and also developed extensive systems to monitor and control audio program loudness. Their system is even more applicable to audio-only transmission systems, and it is already in use in Norway's digital audio broadcasting system.

Figure 48 shows several minutes of jazz music (running left to right) from an actual Internet stream, where 0 dBFS is digital full scale. The chart provides ITU-R BS.1770-2 measurements collected by the Orban meter application:

- Peak level is shown in red;
- Short-term BS.1770-2 loudness is dark blue;
- Loudness target -23 LKFS is shown in yellow;
- Average loudness is light blue;
- Loudness range (LRA) is green; like average loudness, this measurement develops over time, but skips silent intervals; this sample starts at 0 dB, rises and settles around 17 dB on the right.

This jazz sample shows loudness briefly exceeding -23 LKFS, but the long-term average at far right is about -25 LKFS, which is 2 dB below the target. However, peaks exceed the Permitted Maximum Level of -9 dBFS on several occasions, which leaves too little margin to increase program level. The transmission level for this program segment is considered satisfactory as-is.

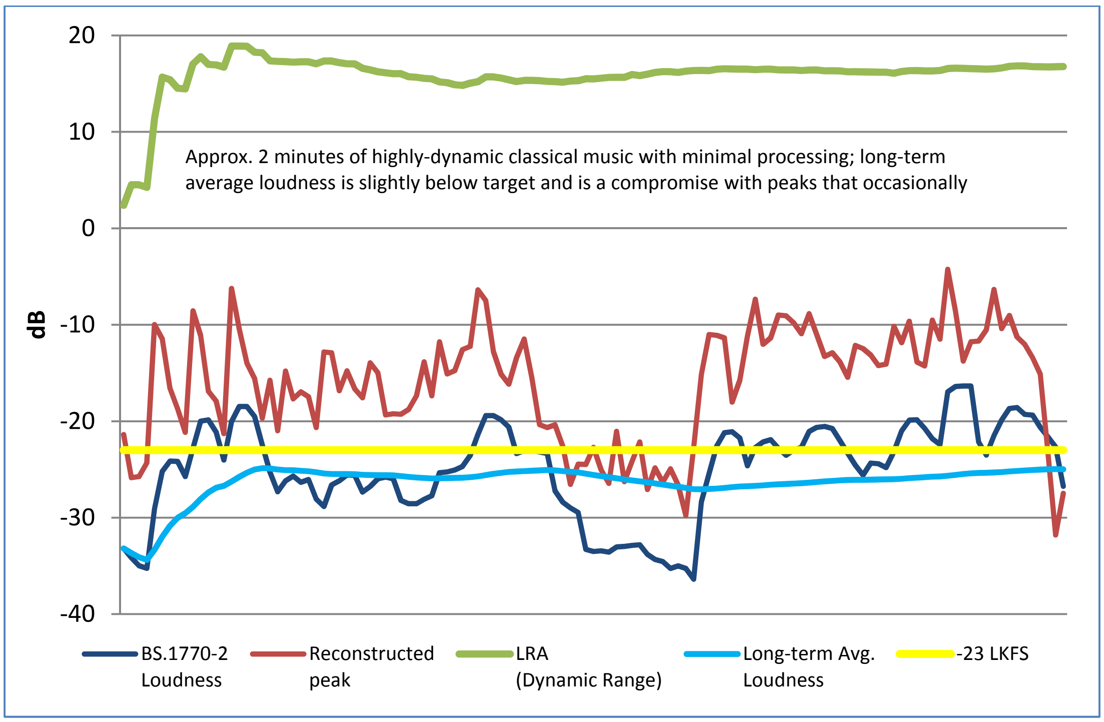


Figure 49 - A 2-minute section of relatively dynamic classical music

A passage of classical music is shown in Figure 49 according to the BS.1770-2 measurements available with the Orban meter. It displays larger variation than the jazz sample, but the long-term average loudness also settles about 2 dB less than the target loudness. Peak level frequently exceeds the Permitted Maximum Level, which argues against an increase in program level. Classical programming may have very wide dynamic range, thus, a long-term average loudness objective of -23 LKFS should be adjusted downward if electrical peaks frequently exceed the Permitted Maximum Level of -9 dBFS.

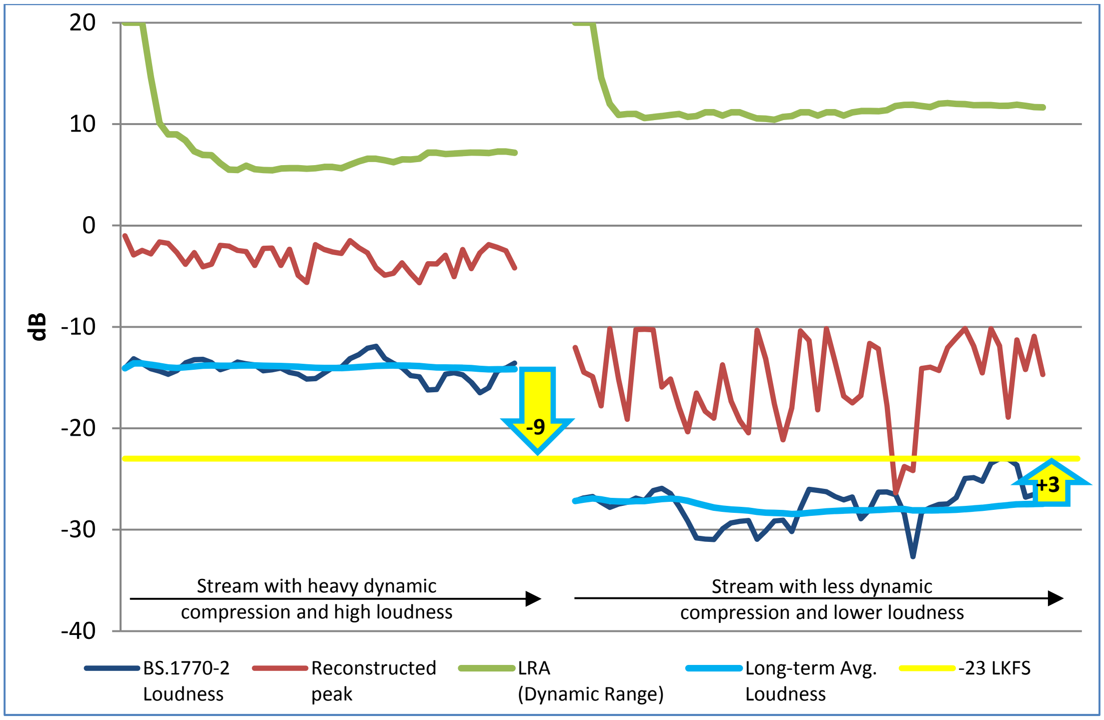


Figure 50 - Recommended ways to handle streams that are too loud or too soft

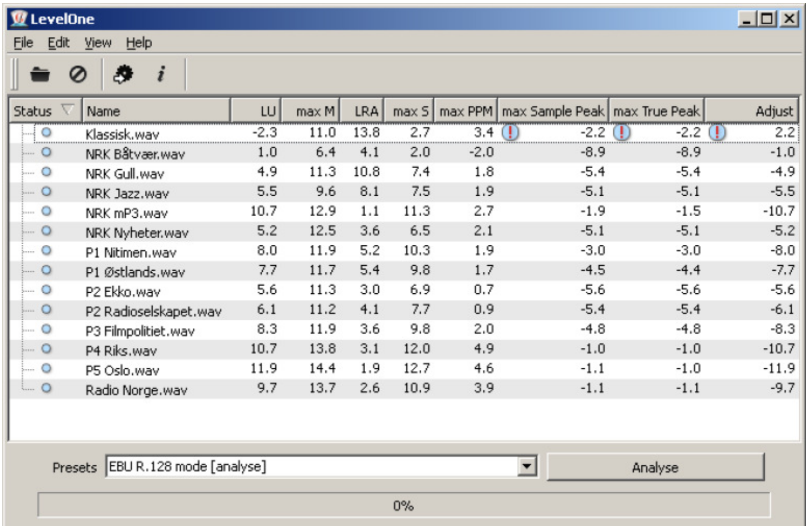
Figure 50 illustrates two actual audio streams and how their transmission levels should be adjusted to provide the best experience for listeners. The stream on the left half is talk with heavy dynamic compression, as shown by its peak level in red. The compression and limiting prevent the peaks from reaching maximum level (0 dBFS), but it also has an average loudness of approximately -14 LKFS, which is 9 dB above the target loudness of -23 LKFS. (Note that the dynamic range, indicated by LRA, in green, is about 8 dB.)

The audio stream on the right is talk with less dynamic compression, with peaks reaching -10 dBFS, and an average loudness of approximately -27 LKFS. This stream could be increased in transmission level by 3 dB without driving peak levels too close to the maximum. For the short interval used for this demonstration, this stream has a dynamic range of approximately 12 dB. While this stream would momentarily vary in loudness because of its more "open" dynamic sound, its long term loudness would be comparable to the denser audio sample on the left, when both are adjusted toward the -23 LKFS target, as shown by the yellow arrows. While the loudness of the first, highly compressed stream is reduced, this approach is the only way to ensure that listeners are not annoyed by jumps in loudness from stream to stream.[16]

[16] There are many of articles discussing the growing concern of the so-called "loudness war" in publicly-distributed audio. Examples of thoughtful papers are: *The Loudness War: Background, Speculation and Recommendations*, Earl Vickers, AES 129th Convention, 4 November 2010; *Loudness normalisation and permitted maximum level of audio signals*, EBU Technical Recommendation R128, European Broadcasting Union, http://tech.ebu.ch/loudness, 2010; .*The Loudness Wars: Why Music Sounds Worse*, NPR Music, 31 December 2009; *Norway's DAB network embraces loudness normalization*, Bjørn Aarseth, EBU Technical Review, 13 February 2012.

BATCH NORMALIZATION OF AUDIO LOUDNESS FOR FILES AND PODCASTS

The job of loudness management, file by file, can be automated with existing software. One example is LevelOne™, developed by Grimm Audio (http://www.grimmaudio.com/pro_software_levelone.htm). This program can serve as part of an ingest system, which, according to Grimm, analyzes files up to 100 times faster than real time, normalizing programs or program segments before transmission or assembly into larger programs. Figure 51 shows the LevelOne window in use by number of NRK program files carried by Norway's Digital Audio Broadcasting system.



Status	Name	LU	max M	LRA	max S	max PPM	max Sample Peak	max True Peak	Adjust
	Klassisk.wav	-2.3	11.0	13.8	2.7	3.4	-2.2	-2.2	2.2
	NRK Båtvær.wav	1.0	6.4	4.1	2.0	-2.0	-8.9	-8.9	-1.0
	NRK Gull.wav	4.9	11.3	10.8	7.4	1.8	-5.4	-5.4	-4.9
	NRK Jazz.wav	5.5	9.6	8.1	7.5	1.9	-5.1	-5.1	-5.5
	NRK mP3.wav	10.7	12.9	1.1	11.3	2.7	-1.9	-1.5	-10.7
	NRK Nyheter.wav	5.2	12.5	3.6	6.5	2.1	-5.1	-5.1	-5.2
	P1 Nitimen.wav	8.0	11.9	5.2	10.3	1.9	-3.0	-3.0	-8.0
	P1 Østlands.wav	7.7	11.7	5.4	9.8	1.7	-4.5	-4.4	-7.7
	P2 Ekko.wav	5.6	11.3	3.0	6.9	0.7	-5.6	-5.6	-5.6
	P2 Radioselskapet.wav	6.1	11.2	4.1	7.7	0.9	-5.4	-5.4	-6.1
	P3 Filmpolitiet.wav	8.3	11.9	3.6	9.8	2.0	-4.8	-4.8	-8.3
	P4 Riks.wav	10.7	13.8	3.1	12.0	4.9	-1.0	-1.0	-10.7
	P5 Oslo.wav	11.9	14.4	1.9	12.7	4.6	-1.1	-1.0	-11.9
	Radio Norge.wav	9.7	13.7	2.6	10.9	3.9	-1.1	-1.1	-9.7

Figure 51 - Example of a commercial batch file processor, made by LevelOne, for file loudness normalization

Another loudness management system is the AudioTools™ Server for file-based workflows (http://minnetonkaaudio.com/). The developer states that this product includes a "dialog anchoring" technique that produces more consistent speech levels, even surrounded by music or sound effects. (These are techniques developed for video production, although they may have application in public radio programs with mixed speech and other sound content.)

An open source solution is FreeLCS (Loudness Correction Server) http://freelcs.sourceforge.net. This free software is set up on a network server where users can drop audio files to be loudness corrected. The software creates graphic files that display the internal loudness variations in a file. Loudness-corrected versions of each file are moved to a "HotFolder" for later transmission. The program supports encoded file formats and uses a protective limiter to prevent peak levels from clipping where the volume must be increased. The software is written in Python 3 which allows modifications and integration with other programs and runs on Ubuntu Linux.

LOUDNESS LEVELS RELATIVE TO OTHER INTERNET STREAMS

The Internet is open territory as far as stream loudness is concerned. An increasing number of stream operators are compressing their program content and operating peaks close to maximum level. Even in two of three streams of the BBC, dynamic compression is evident: Figure 52 shows sample clips from Radio 3 (classical), Radio 2 (adult contemporary) and Radio 1 (Top 40). The vertical scale is dB relative to digital full scale (for the red and blue traces) and the ITU BS.1770-2 loudness scale, dBLK, for the green trace.

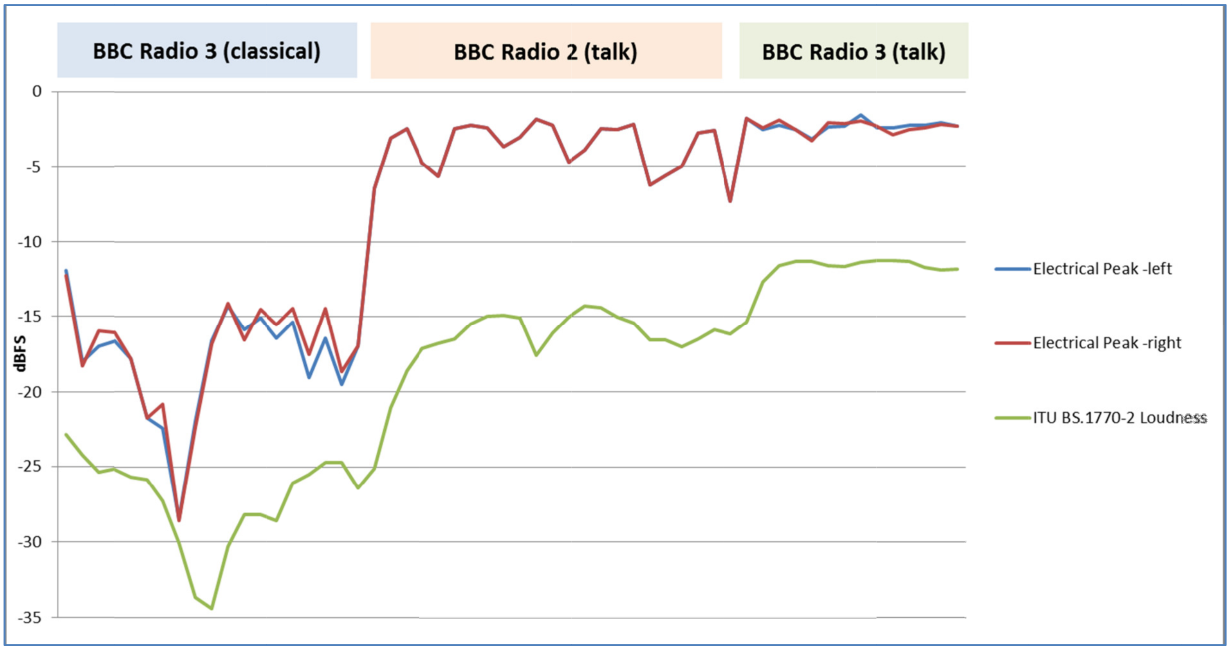


Figure 52 - Samples from BBC Internet audio streams: Radio 3, Radio 2 and Radio 1 (1 minute total)

In this case study, Radio 2 and 3 were engaged in talk during the 15 to 20-second samples. The classical music at the time was a light orchestra rendition, with peaks just below -15 dBFS and loudness of about -35 to -25 dBLK. This approximates the loudness of a public radio stream, transmitted at the level standards described earlier (Permitted Maximum Level of -9 dBFS and target loudness of -23 dBLK).

By comparison, the Radio 2 and 3 segments are compressed, raising electrical peaks within 2 dB of full scale, and loudness in a tight range between -15 and -11 dBLK. Of course, they sound substantially louder than the Radio 3 program, which raises an issue as to whether public radio should compete with loud streams.

Active loudness management in the stream players is recommended, by putting a limit on program audio that exceeds a nominal level (the recommended loudness in EBU R128 is -23 dBLK). In this case study, classical music would be unaffected by a threshold at -23 dBLK, but the Radio 2 and Radio 3 programs would be “turned down” by the loudness management software in the player. The software may also be programmed to reduce the range of program material with wide variations while also preventing “blasting” of loud program material (streams that are not compliant with EBU R128).

If transmission level must be raised, and lacking loudness management on the consumer side, a process that has the least impact on program dynamics is to: (1) produce to target level (-23 dBLK) as usual; (2) limit peaks at -9 dBFS; and (3) apply a gain increase of ”n” dB (n << 9 dB).

STREAM SERVER

The computer supplied to stations for stream encoding is a SuperChassis 512L-260B, made by Super Micro Computer, Inc. (http://www.supermicro.com/products/chassis/1u/512/sc512l-260.cfm). It is a 1RU rack mount unit running an Intel® processor and C602 chipset with 4GB of memory and two 1TB hard drives. The motherboard expansion slot supports one PCI-E 3.0 x 16 full-height half-length card, which lays horizontally because of the low form-factor of the cabinet.

PRESENT AUDIO CARD

For audio input, NPR Digital Media selected an E-MU®-0404 PCI audio card made by Creative Technology Ltd. (http://us.store.creative.com/EMU-0404-PCI/M/B00064YZVK.htm). The card is one of the highest performance products developed for the consumer audio market, using 24-bit 192 kHz A/D converters and DSP hardware acceleration effects to minimize CPU load. As a consumer product, the card, shown in Figure 53, is supplied with dongle adapters that interconnect the card frame to standard consumer connectors. The analog (DB-9) and digital (DB-15) audio connectors are detailed in Figure 54.

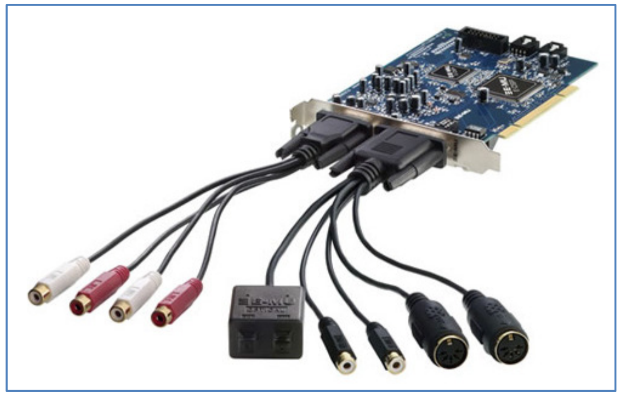


Figure 53 - E-MU 0404 card with dongles attached

The audio lines extend to unbalanced input and output connectors operating at -10 dBu. Digital input and output is extended to optical and unbalanced coaxial operating at S/PDIF signal level (0.5 V p-p). The product is furnished with PatchMix DSP™ software for Windows computers that provides a comprehensive array of audio mixing and processing effects.

The consumer analog and digital levels can present a problem for stations, which typically operate balanced analog audio lines at substantially higher levels. Professional users must also convert from balanced to unbalanced audio at the card to avoid hum and noise. These conversions also increase difficulty with standardizing transmission levels and providing proper headroom at the card's A/D converter. Also, stations use professional connectors, such as XLR to interface analog or digital audio equipment. Stations also may use AES-EBU digital lines to interconnect audio equipment. While the protocol for S/PDIF and AES digital audio are compatible, AES operates at a higher voltage and is usually a balanced connection. These differences could lead to reliability issues with the audio interface.

NPR Labs investigated a variety of professional audio interface cards to replace the EMU-0404, but the cost would be far beyond what was budgeted, and NPR had already purchased a large number of the consumer cards. As a solution for stations and NPR, NPR Labs recommended an interface box be supplied to stations. No off-the-shelf box was available, so we designed a custom box built locally by Excalibur

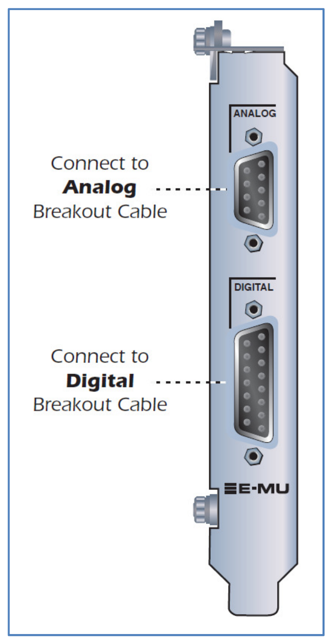


Figure 54 - Audio connectors on card bracket

Electronics. This box provides a pair of female XLR connectors for the audio input and an RJ-45 connector for the output, which is used with a cable and DB-9 connector for direct connection to the E-MU card's analog input.

Figure 55 - Custom audio interface

As shown in Figure 56, the box contains two quality Triad Magnetics® transformers and a resistive pad using metal film resistors. The analog path is designed to produce a digitized level of -18 dBFS with a 50 ohm source impedance producing +4 dBm. (As discussed in the section on transmission levels, -18 dBFS is the nominal transmission level for streaming, and this also provides 18 dB of headroom for the A/D converter's full-scale point.) The pad impedance is designed to present a bridging load to the balanced source which reduces the level by less than 2 dB if sourced from 600 ohms. Internal jumpers are provided to optionally connect the input and output connectors' ground terminals to the metal box. As shipped, the jumper on the XLR inputs is lifted from the chassis ground; the jumper for the ground connection to the E-MU card is connected. To avoid ground loops it is recommended that only one jumper be in place.

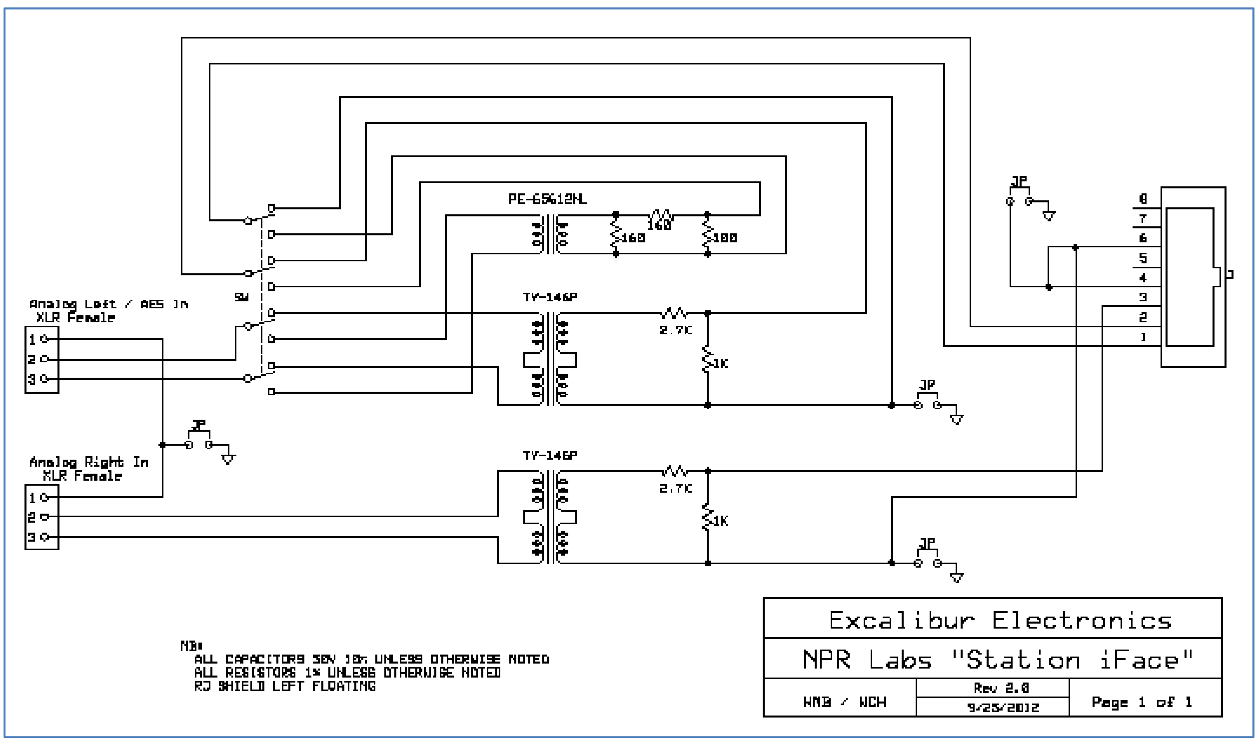


Figure 56 - Schematic for audio interface box

A rear-panel switch is provided so that the left XLR connector can serve as the AES-EBU digital input. The digital signal is connected the RJ-45 output connector, which can be connected through a separate cable to the E-MU card's DB-15 S/PDIF input connector. Tests were made to verify that the card works reliably with AES-EBU signals (minimum 1.2v P-P, normal AES_3 operates at 2-3 v P-P).

The lowest sample rate for the preselected audio interface card (Creative E-MU 0404) is 44.1 kHz, which is the recommended rate for music. If it were available in the present configuration, we would recommend a rate of 32 kHz for streams with primarily news and information programming. This would avoid quantization of audio spectrum that is unnecessary for this genre, which would provide a small improvement in codec efficiency. However, if a future codec with a vocoder is employed, this sample rate change may be unneeded.

PATCHMIX DSP CONFIGURATION

The Creative E-MU cards require proprietary software, called “PatchMix DSP” to operate. When installed, they software bypasses the standard Windows audio mixer and provides a digital PCM audio connection directly stream encoding software as “Line In (Legacy Mode)”. The operation of the Windows audio mixer, the Realtek ALC662 audio chip on the motherboard, and its interface software are unaffected; they may be used simultaneously as a second audio input or as an audio bridging monitor of the E-MU PCM output, supplied to the 3.5mm jack on the rear panel.

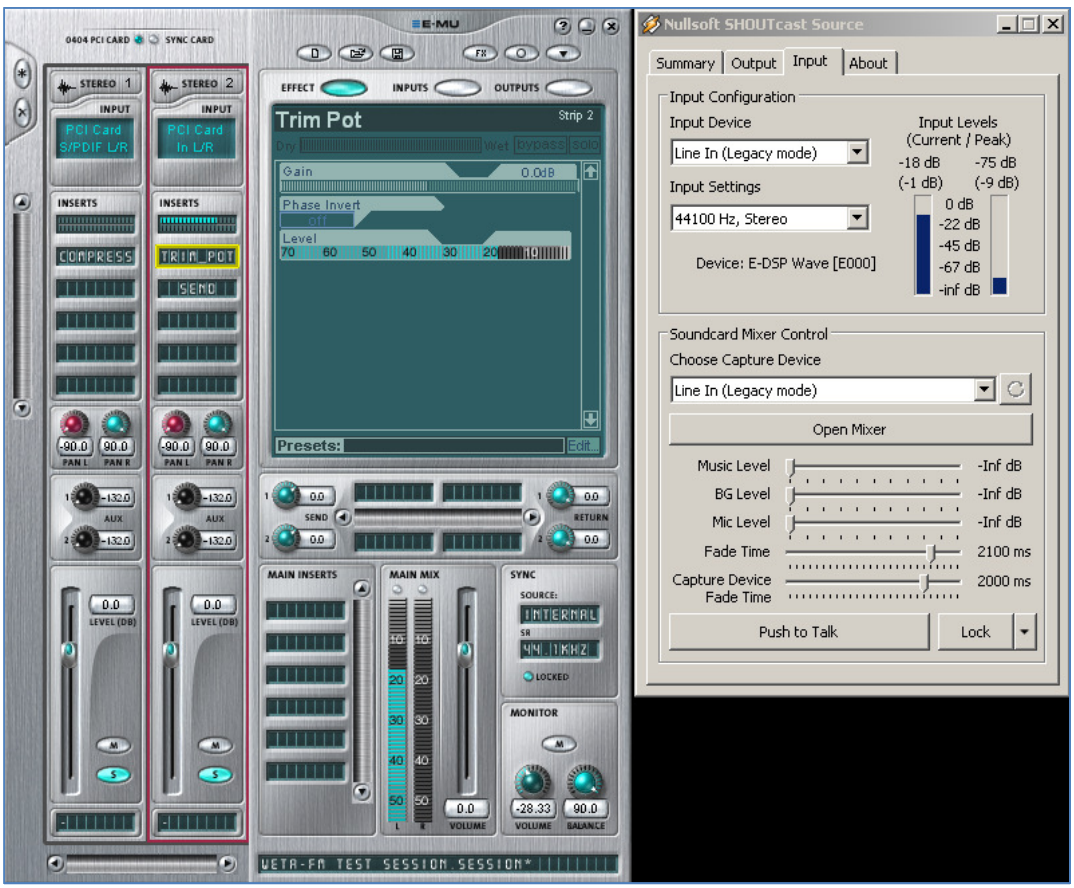


Figure 57 - PatchMix DSP window (left) and Winamp SHOUTcast window (right)

The PatchMix DSP software loads automatically at startup, along with the Winamp player and its associated SHOUTcast DSP plugin, which provides the stream encoding. (This applies only to streams not using the Triton Digital “Ad Insertion” streaming software.)

Figure 57 shows the recommended configuration for PatchMix DSP, with two input "modules": one for S/PDIF digital input and one for analog L/R input on the E-MU 0404 card. The modules provide six inserts each for special features. The S/PDIF module's top insert is a small bar graph peak level indicator, followed by a DSP-based COMPRESSor/limiter. The analog is also configured with a bar graph indicator, followed by a TIRM_POT gain control, and then a SEND insert, which routes the module's output to the computer's PCM audio feeding the encoding card. Only one SEND can be inserted, which serves as a 'toggle switch' for either digital audio input or analog input. The TRIM_POT insert was clicked, which put a yellow box around it and displays information about the insert: in this case a larger bar graph indicator in the "video" screen.

The TRIM_POT screen includes a gain control, which is set to 0.0 dB. In this case, a +4 dBu input on analog left channel produces -18 dBFS on the video screen's bar graph indicator. The SHOUTcast encoding software is set to receive the PCM audio from PatchMix ("Line In"), which produces a -18 dB indication on the encoder's bar graph indicator. Note that maximum peaks would occasionally reach -9 dB on the encoder, which provides 9 dB of headroom. This is the nominal level setup for transmission.

TRITON DIGITAL AD-INSERTION SOFTWARE

To support an audience measurement and ad-management solution for public radio streams, NPR plans to work with Triton Digital (formerly Ando Media) on a digital sponsorship system. The system would use the stream server hardware, described above, along with Triton media automation software to insert sponsorship announcements.

The Triton documentation specifies that two sound cards are required: one for the external audio input and a second to provide the internal announcement files.[17] However, the SuperServer computer selected by NPR physically supports only a single PC card, such as the E-MU 0404 audio soundcard.[18] The Realtek ALC662 audio chip on the computer motherboard can provide the announcement file playback.[19] As the announcement files are played digitally, the audio is passed digitally through the Realtek chip and no degradation in quality occurs. However, there is no provision in the Triton software to automatically adjust the loudness of recorded announcements. This is a potential problem for uniform transmission loudness, as the announcements are produced elsewhere under differing level standards, possibly including dynamic compression. Ideally, Triton should upgrade its software to adjust announcement playback levels to match the external (station) audio, in accordance with EBU R128. Lacking that, implementation of active loudness management in the playback

---

[17] *Ando Media Installation Client Requirements and Preparation*, Ando Media Group, March 2, 2011.

[18] We understand from Triton engineers that their ad-insertion software was written with the intention of using the Orban Optimod PC 1101 audio card (now the Optimod-PC 1211e), which supports external and ad-insertion audio. This is a relatively large increase in cost for the stream server and, if on-card audio processing is used to control announcement levels, would also dynamically compress the station program audio, which we discourage.

[19] We have tested the on-board chip for operation with the E-MU card, however, we did not have access to the announcement files to verify operation the Triton ad-insertion software. We are advised by Trition's technical support that the motherboard audio chip should be compatible with their software.

software, as discussed earlier, can effectively moderate undesired changes in loudness as announcements are inserted. Currently, the Triton software also can support only one stream per server, which prevents dual-streaming at different bit rates or with different codecs.

AUDIO CARD REPLACEMENTS

The E-MU 0404 product is discontinued, and Creative has no direct replacement for the model. NPR Digital Media followed our advice and purchased some remaining cards at dealers. However, there is a need to identify a long-term replacement for this card as additional computers are implements or old cards fail. The Optimod-PC 1211e card is a candidate, which offers the capability to mix the external audio input and the internal audio for Triton "ad insertion" software. (Triton officials noted that their software was written with the intention of using the Optimod-PC card, which also includes a professional audio compressor/limiter.) However, the Optimod-PC card costs several times more than the E-MU card, even with the audio interface.

The search for a solution to the present card's consumer input gave us a perspective on the direction of PC audio interfaces. In our view, audio engineers are moving away from PC card implementations, and it was apparent that the choices of these cards has quickly diminished over the past couple of years, in favor of external audio interfaces with USB interconnection to the PC. USB audio interfaces are quite plentiful, have very high audio quality and could permit more than one audio input to the computer.

There is some concern about whether USB devices are reliable enough, especially when the computer is rebooted: does it automatically restart the audio interface every time? Also, questions have been raised about the possibility of interruptions to the audio data through the USB port as other devices use the port or computer's hardware data bus. These are questions on which we recommend further study for future growth, and before the supply of E-MU PC cards runs out.

CAUTIONS FOR THE SOFTWARE ENCODER

The stream encoding software selected by NPR Digital Services is the Nullsoft's Winamp Standard media player with the SHOUTcast DSP plugin for stream encoding. This software, currently in version 5.63 for the player, is available as a free download at www.winamp.com. There are some uncertainties with using freeware for a large-scale streaming project such as NPR's:

- Correcting errors in the software, or answering difficult questions may be difficult without technical support that is under contractual obligation to NPR;
- The software is designed for use by individual users who undertake their own local downloads, which may make updates to the software across hundreds of remote computers more cumbersome;
- Modern audio codecs are under continual improvement, and new versions are issued from time to time that should be used; the updates may not be apparent to NPR and more difficult to push out to large numbers of remote computers.

Fortunately, Winamp has a large user base and online community. (There is a "Pro" version of the software that promises technical support for customers, but a question emailed to them has remained unanswered after 10 days.) It is possible that others will identify software issues quickly, which technical support may correct on behalf of the community, and some technical questions may be answered by experts in the community.

ACKNOWLEDGEMENTS

The authors appreciate the assistance and contributions of the following individuals and companies:

Jan Nordmann, Max Neuendorf, and Nikolaus Rettelbach, Fraunhofer Institute
Claude Gravel, VoiceAge Corporation
Towson University
Bob Orban and Greg Ogonowski, Orban
The members of the Codec Evaluation Project Advisory Group