DIALNORM_TUTORIAL
DATE: 28 MAR 2010

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DIALNORM

A Tutorial from the Perspective of the Motion Picture Industry will be presented. Why the Motion Picture Industry? The audio standard for terrestrial digital TV was lifted from the multi-channel audio format used for the distribution of motion pictures to your local neighborhood theater and on commercially produced movie DVD's.

LET'S ALL C.AL.M. DOWN

A bill to regulate the volume of TV commercials (apparently not radio commercials or movie trailers) is working its way through the House and Senate. Passage is expected and Congress is expected to recommend that the ATSC Recommended Practice: "Techniques for Establishing and Maintaining Audio Loudness for Digital Television" (soon to be released by the ATSC) is the basis for determining compliance. This document is basically a rewrite of the ATSC document A85 and includes several practical application examples.

THE ANSWER

Since Engineers like to skip to the end of the murder mystery book to find out who the killer is, I will jump to the end of the presentation and give you the answer first. When the Advanced Television Systems Committee wrote the technical specifications for over-the-air digital television broadcasting, they included as part of the audio specification, a mystical entity called "Dialnorm". This mystical entity was intended to control the volume of the program audio at the consumer end of the broadcast chain. The Dialnorm metadata would control an audio attenuator in the consumer TV.

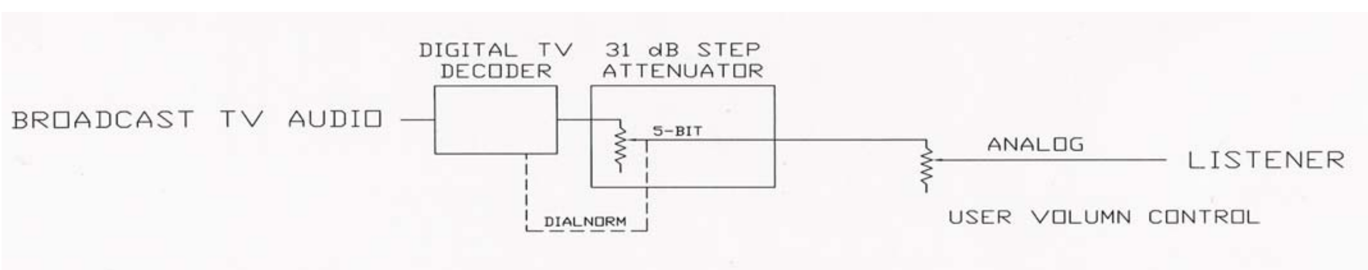

Figure 1 – Dialnorm Implementation

The theory is that if the program audio is too loud, the Dialnorm data would "turn down" the volume automatically. The key point to remember is that Dialnorm is only present in Dolby Digital or Dolby-E digital audio streams. There is no provision to carry Dialnorm data in analog audio or AES digital audio.

If Dialnorm is the answer, let's figure out what the question is.

WHAT IS "LOUDNESS"

We listen in three general modes- Casual, Informative and Emotional. Casual is listening at a level below normal conversational level, usually for background music. The brain can go into "casual" mode if we are not actively participating in the event. Informative listing is at the normal conversational level where we are intent on understanding the audible event. Emotional listening occurs when we enjoy a musical performance at performance levels, or when someone is yelling.

How we judge loudness depends on our mood, personal preference and how much we like the audible performance. For example, I like classical music at 90 SPL, any county and western music over 30 SPL I find annoying! (no offense- I'm sure many of you have similar emotional preferences, I put opera in the 30 SPL bin).

When it comes to judging loudness, people are in fairly close agreement when it comes to dialog, perhaps because we are used to conversational levels. We tend prefer our music louder than dialog by about 10 dB. We also tolerate dialog that is too soft verses dialog that is too loud by a 2:1 ratio. Dialog that is "too loud" is akin to someone yelling, which most people find annoying.

With the advent of digital broadcasting, we can no longer rely on the use of "modulation" monitors to ensure compliance with FCC Rules regarding the "100 %" modulation limit. Traditionally, analog broadcasting used fairly aggressive audio limiting and compression which resulted in a restricted dynamic range. Digital audio has a dynamic range twice that of analog and a noise floor well below that of any of the analog audio gear currently available. The challenge we face is whether we can use this increased dynamic range, and if we do, will we run afoul of the C.A.L.M. Act?

A DEMONSTRATION OF PERCEIVED LOUDNESS

Loudness can be described in several ways, as the following list indicates.

Objectively- Scientific measure of SPL
Subjectively- Personal preference, density, active channels
Perceived- How the ear/brain can mislead
Emotional- How we feel about the source material

The C.A.L.M. act is not concerned with objective measurement but the subjective and emotional loudness levels as perceived by the listener. Dialnorm is an attempt to quantify these subjective and emotional issues. Phillips has produced an audio CD of auditory demonstrations for the Institute for Perception Research of Eindhoven, The Netherlands. The CD has 80 tracks that demonstrate audio perception illusions. The one track that will be demonstrated is Track 7- "Demonstration 3. Critical Bands by Loudness Comparison. The test information is given in Figure 2- "Loudness verses Bandwidth"

(Spoiler Alert- As you listen to the test sequence, as the bandwidth increases the perceived loudness decreases slightly until the critical bandwidth is reached, when the perceived loudness increases.)

The Phillips Audio Demonstrations is available from Old Colony Sound Lab (www.audioxpress.com), item CDAC for $37.95 plus shipping. This CD is highly recommended if you are interested in understanding how the ear-brain combination can be tricked by everyday audio presentations.

Demonstration 3. Critical Bands by Loudness Comparison (1:09)

This demonstration provides another method for estimating critical bandwidth. The bandwidth of a noise burst is increased while its amplitude is decreased to keep the power constant. When the bandwidth is greater than a critical band, the subjective loudness increases above that of a reference noise burst, because the stimulus now extends over more than one critical band.

The subjective loudness of a complex tone is fairly complicated, but for combining the loudness of two or more tones, the following rules of thumb usually apply:

1. If the frequencies of the tones lie within the critical bandwidth, the loudness is calculated from the total intensity: $I = I_1 + I_2 + I_3 + ...$
2. If the bandwidth exceeds the critical bandwidth, the resulting loudness is greater than obtained from a simple summation of intensities. As the bandwidth increases, the loudness approaches (but remains less than) a value that is the sum of the individual loudnesses: $S = S_1 + S_2 + S_3 + ...$

In this demonstration, a noise band of 1000-Hz center frequency and 15% bandwidth (930-1075 Hz) is followed by a test band with the same center frequency and bandwidth (see *1* in the figure below). The bandwidth of the test band is then increased in 7 steps of 15% each, while the amplitude is decreased to keep the power constant. When the bandwidth exceeds the critical bandwidth, the loudness begins to increase.

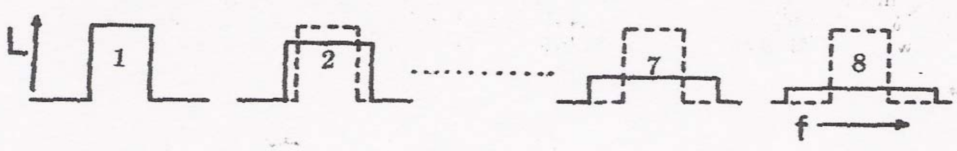

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Figure 2- Loudness verses Bandwidth

DYNAMIC RANGE & SPL LEVELS v.s. RECORDING FORMATS

Here is a list of typical sound pressure levels that we typically encounter in our daily lives.

0	SPL	threshold of audibility (20 uPa) [eardrum movement appx width of Hydrogen molecule]	
<30	SPL	eerily silent	
30	SPL	soft whisper at 5'	
30-40	SPL	typical broadcast or recording studio	
40	SPL	typical home environments	
50	SPL	typical office	
55	SPL	background music	
65	SPL	causal listening level	
70	SPL	average conversation level at 1'	
70-75	SPL	average consumer listening level	
75	SPL	small chamber group	
80	SPL	piano at 20'	(0 level ref for television work)
85	SPL	concert hall seat	**(0 level ref for film work)**
94	SPL	1 Pascal (1 Newton/ SQ Meter) eg- about the weight of an apple	
95	SPL	small jazz band at 20'	
105	SPL	wood working shop	
112.1	SPL	1 acoustic watt radiated by 100% efficient speaker @ 1 m, half-space	
115	SPL	large orchestra- conductor's podium	
> 120	SPL	hearing protection	

Table 1 – Loudness in terms of Sound Pressure Level

SPL falls at the rate of 6 dB for each doubling of distance from the source. The Sound Pressure Level, SPL, is one of the fundamental quantities used in describing sound reproduction systems. The goal of any Hi-Fi system is to reproduce naturally occurring sound levels. SPL is expressed in dB referenced to the threshold of audibility, defined as 20 uPa. SPL is technically amplitude not a level measurement. SPL is always a positive number as far as we are concerned, negative SPL's are used to describe the absorption of sound.

The dynamic range available in any recoding format is bracketed at the upper end by the maximum operating level (MOL) and at the lower end by the systems noise floor. We are all familiar with the noise floor limitation and the MOL of analog systems (tape surface smoothness, tape erasure level, disk surface noise, film grain and electronic noise floor verses tape sat, disk grove cuts, film exposure limits and electronic clipping).

Digital recording defines dynamic range (ie- the SNR, Signal to Noise ratio) in an entirely different way that is independent of the recording medium, SNR is defined by the number of bits available, 16-bit digital audio gives a nearly 100 dB SNR. [$SNR = 6.0206 \times n + 1.7609 = 98$ dB]

The listening environment also affects the available dynamic range; room noise (dishwasher, traffic, kids, wife) limits the low end verses at the upper end, disturbing the neighbors.

At this point, we need to introduce the concept of "headroom". Headroom is the distance between the nominal operating level and the MOL. Since music is dynamic, unlike a sinusoidal test tone which has a mathematically defined average (63.6% pk), nominal (70.7%) and peak level; music has a nominal level with peaks of 14-17 dB above the nominal levels.

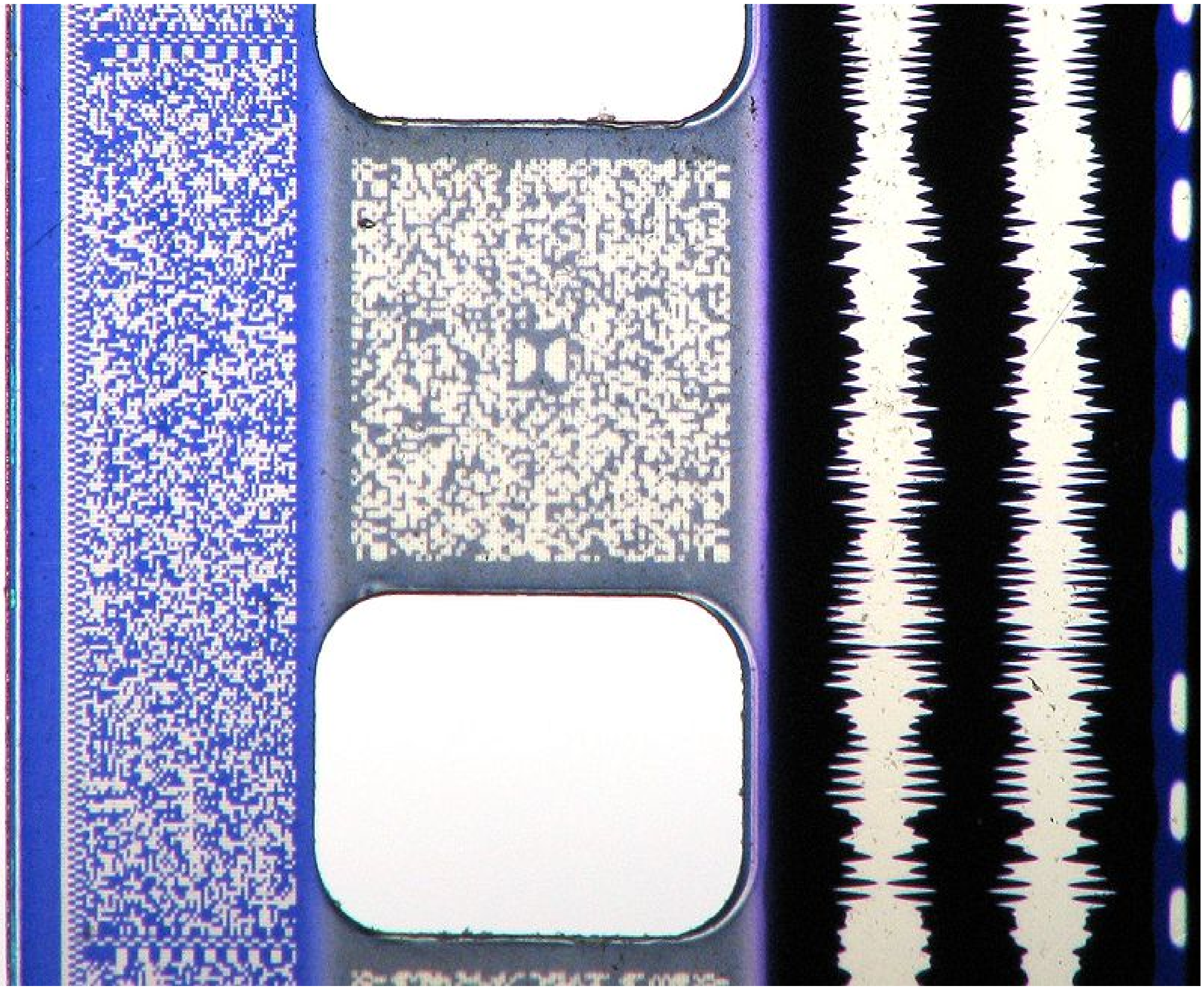
In digital broadcasting, we define the "nominal" level (ie- the 0 dB sine wave (peak in this case) test tone reference level) as -20 dBFS (nominal equals 20 dB below digital Full Scale, that is all audio bits set to one). This more than meets the requirement of 14 to 17 dB of headroom for music.

ENTER RAY DOLBY

The motion picture industry has suffered with an extremely poor audio recording format known as optical audio. The optical format is noisy and has a high-frequency roll-off that begins at around 2 KHz and is approximately 30 dB down at 16 KHz; this is the Academy mono format.

Ray Dolby started life in the record industry- he introduced a method of noise reduction for audio recordings, called Dolby Type-A. Later he introduced a cost reduced version for consumer tape formats called Type-B. Dolby went on to apply Type-A noise reduction to the optical sound tracks of motion pictures, which lowered the noise floor by about 10 dB and flatted the frequency response to 8 KHz and 'only' 10 dB down at 16 KHz. When cinema went wide screen, the need for more audio tracks evolved. Dolby created a way of encoding 4 channels of audio on two optical film channels, calling it Dolby Stereo, which we know as Dolby Surround (technically speaking, Dolby Surround is a L-C-R format). Dolby Stereo is backward compatible with the existing installed base of optical stereo theaters and even the mono optical readers. Dolby then introduced the SR noise reduction format for the optical film tracks which gave another 3 dB improvement in film audio SNR and further flattening the response to 16 KHz.

The advent of digital required Dolby to re-invent them selves. Dolby Digital was introduced to the film industry to compliment the optical sound (see Figure 3 – "Optical film audio distribution formats"). Dolby Digital supports up to six audio channels. Dolby sat on the ATSC committee, and proposed Dolby Digital as the audio format for DTV, which the ATSC accepted and incorporated it into the ATSC DTV Standard.



←PICTURE [SDDS DIGITAL][DOLBY DIGITAL][STEREO OPTICAL][DTS]

Figure 3 - Optical film audio distribution formats

The film picture image is off to the left, this view is from the projection booth looking toward the screen. SDDS is the Sony Dynamic Digital Sound format. The area between the sprocket holes is the Dolby Digital format data (note the DD logo). The standard stereo optical tracks (which may be Dolby Surround encoded) is between the sprocket holes and the edge of the film. The vertical dashes near the edge of the film is Time Code that locks the film image to an external CD player that caries the Digital Theater System audio. Most theatrical release prints carry all four audio formats to accommodate the varied types of installations encountered. Many digital playback systems revert to the optical tracks if sprocket hole damage or other film defects render the digital data temporally unreadable. All theaters are capable of optical playback and usually have at least one of the digital playback systems.

TWO PROBLEMS WITH DIGITAL AUDIO

The increased dynamic range that digital audio provides is incompatible with conventional TV viewing, which has to overcome room noise distractions and the potential for annoying other members of the household. The increased dynamic range has also led to inconsistent dialog audio levels, with film work recording dialog at -30 dBFS and broadcast recording dialog at -20 dBFS. In actuality, broadcast dialog levels seem to be anywhere from below -20 dBFS to approaching 0 dBFS! Dolby introduced two solutions to these problems, Dial Norm and Dynamic Range Control (DRC). To the best of my knowledge, Dialnorm and DRC are not part of the Dolby Digital film standard; let me know if this is a false assumption.

DIALNORM

In film work, using digital audio, it was generally agreed that dialog levels were consistently running 30dB below 0 dBFS, or -30 dBFS, in other words, the dialog should be recorded at a level of -30dBFS, giving film audio 30 dB of "emotional" headroom. Since it was anticipated that dialog may be recorded hotter in commercial and television production, a method of normalizing the audio level to the film standard of -30 dBFS, hence the term Dialnorm (dial for dialog, norm for normalization), for dialog normalization..

Why -31 dBFS for the reference Dialnorm level? To get a 30 db digitally controlled attenuator, we need a minimum of 5 data bits, which gives us a 31 dB attenuator, hence the reference level of -31 dBFS. Dialnorm can be any value from -31 (no gain reduction) to -1 (30 dB of gain reduction). A Dialnorm value of zero is allowed (default value of most authoring programs), and is interpreted as -31 (no gain reduction)

Attenuation = 31 + (Dialnorm value)

EXAMPLE: 31 + (-20 Dialnorm value) = 11 dB of attenuation [01011]

Now that we know how Dialnorm controls the attenuator to adjust the program level at the consumer end of the chain, how is the Dialnorm value determined? The Dialnorm value is inserted in the ATSC AC3 audio signal just before transmission. The incoming AC3 Dialnorm value, which is assumed to be present and correct, should be passed on to the consumer. In reality, most broadcasters insert a fixed value of Dialnorm in the AC3 encoder. It is assumed that the inserted Dialnorm value matches the program dialog level. The Dolby LM100 is a piece of audio test equipment that can determine the level of the dialog in any analog, AES or AC3 (Dolby Digital) audio signal. This measured level is then the Dialnorm value. For example, if the dialog averages -20 dBFS, the Dialnorm value is -20, and the digital attenuator in the consumer receiver inserts 11 dB of attenuation.

Dialnorm is intended to ensure that all audio levels are the same, and if not, use the Dialnorm value to automatically control an attenuator in the consumers' receiver to make them the same. Effectively, program audio level adjustment has been moved from the audio operator, who should be maintaining a consistent audio level between programs, to a semi-automated system that determines the degree of program audio level variation between programs and applies the required correction factor at the consumer end of the audio chain. In other words, it is now assumed that we will transmit varying audio levels, both within our station and between other stations, and perform the level correction in the consumer receiver. This whole premises depends of the Dialnorm value being transmitted, accurately represents the actual audio level. At the end of this presentation I will show some off-air measurements that reflect the reality of the current status of Dialnorm.

DYNAMIC RANGE CONTROL

DRC is a method to fit the 100 dB dynamic range that digital audio gives us into the typical 50 dB consumer listening environment. Figure 4A shows how dynamic range compares to typical source levels with the loudest source normalized at 0 dBFS (compare with Table 1) and Figure 4B shows typical levels with respect to film levels. Two key points are evident, in Figure 4A we note that the typical listening level is below that of live performance levels and from Figure 4B, the dynamic range digital offers can not be realistically expected in the home listening environment.

Figure 5 shows the relative dynamic ranges available in the analog and digital recording environments.

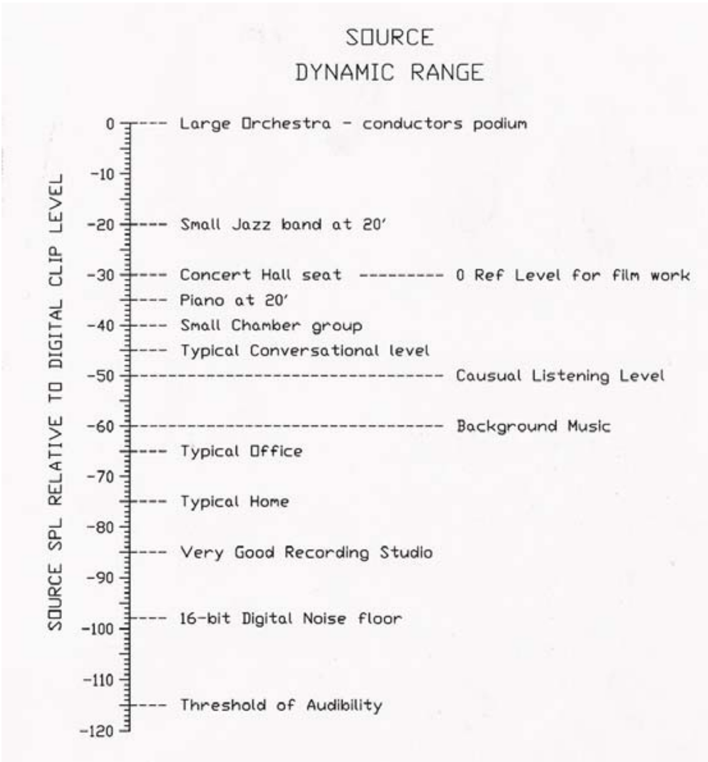


Figure 4A

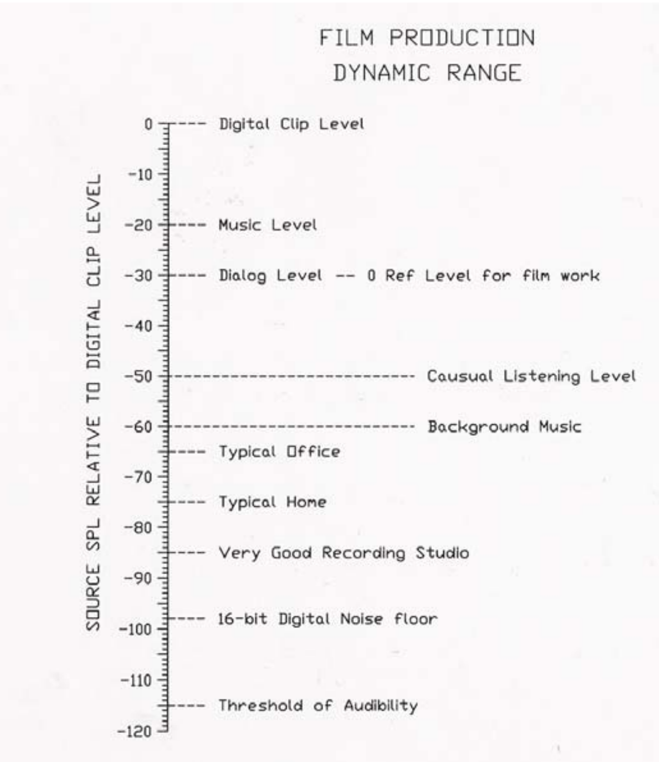


Figure 4B

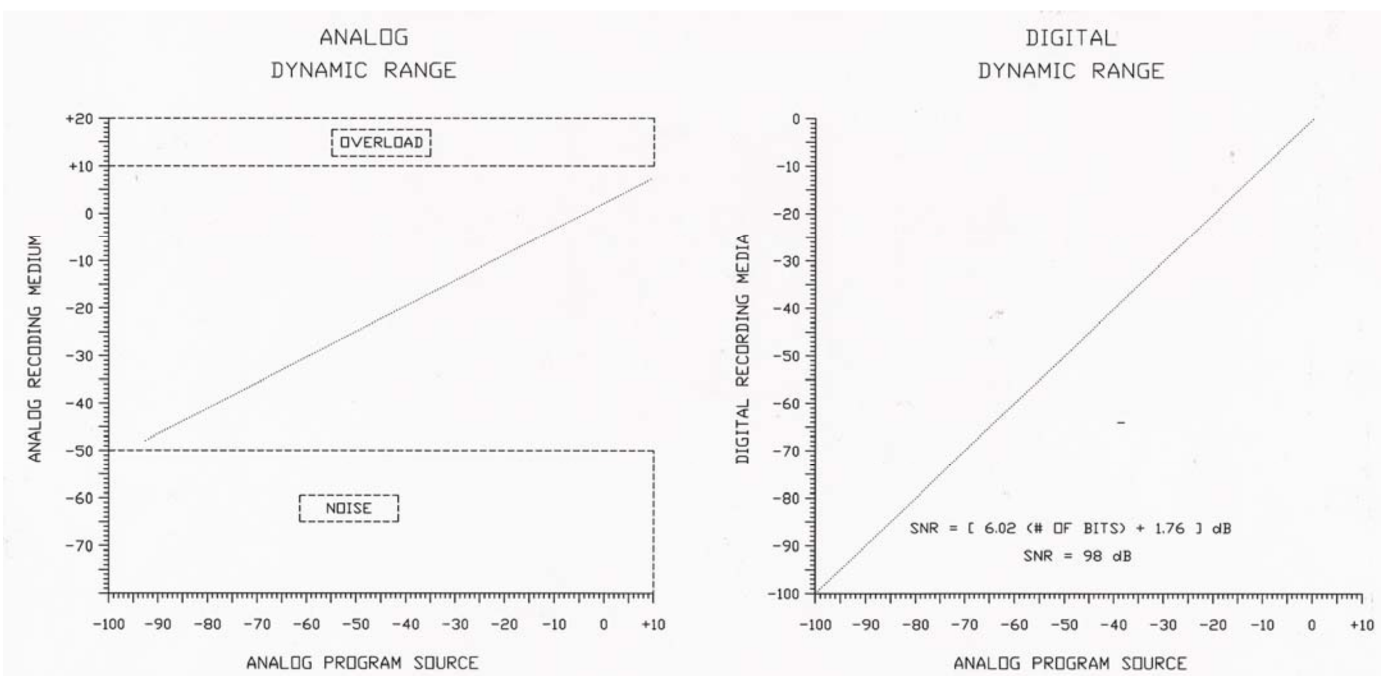


Figure 5 – Comparison of analog and digital recording dynamic range

DRC is a playback compression system that is part of every digital AC3 receiver. There are 5 preset profiles as shown on Figures 6, 7 and 8. There is a sixth preset under user control called "off". Most Home Theater Receivers dumb down the user DRC choices to OFF, LOW, MID, HIGH, where HIGH is the so called "MIDNITE" mode.

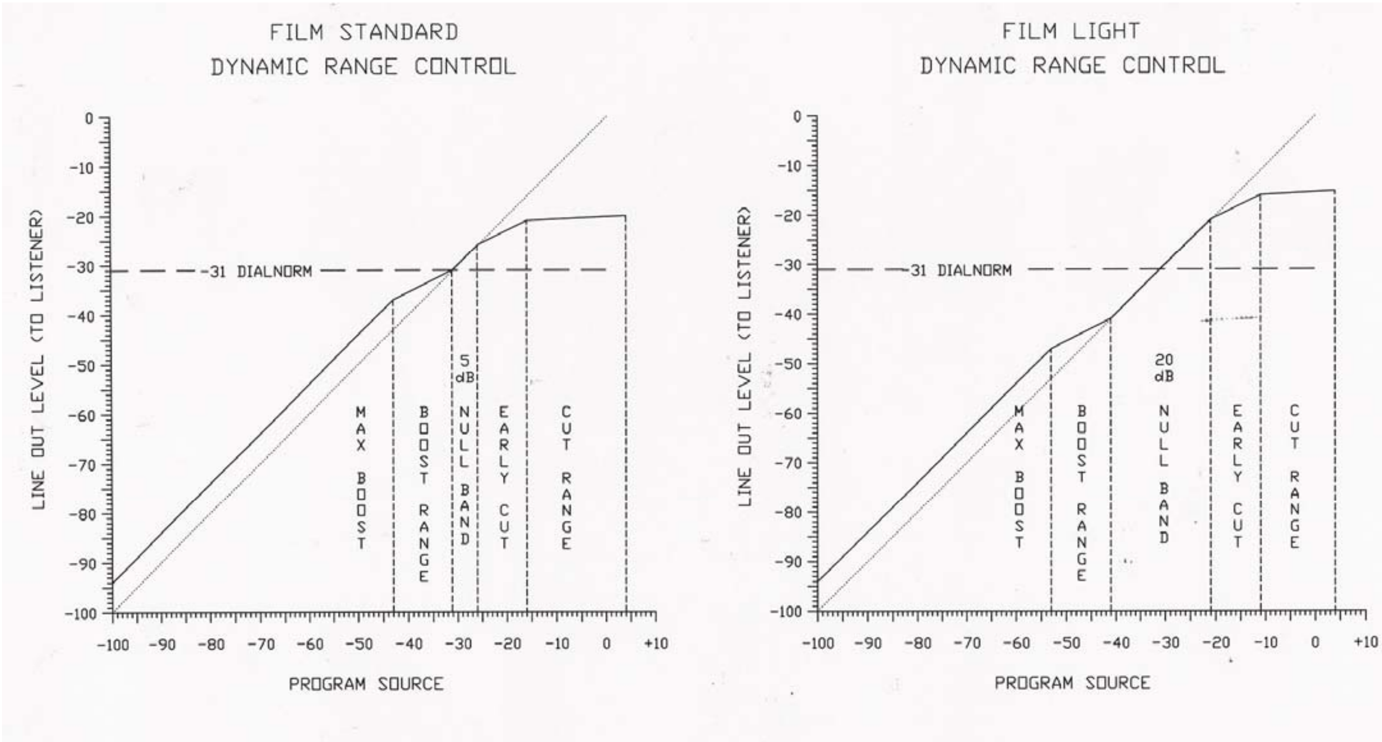


Figure 6 – DRC settings for film (DVD and DTV)

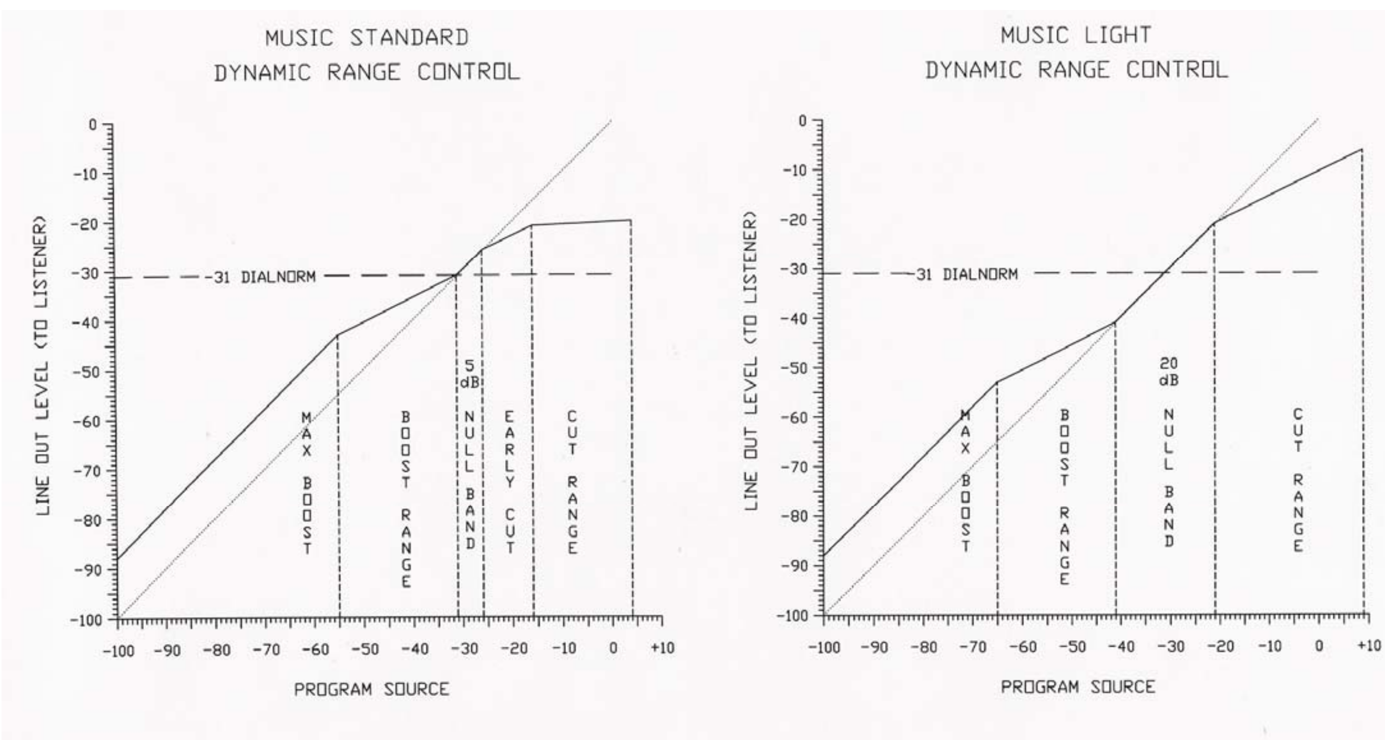


Figure 7 – DRC settings for music (non-dialog material)

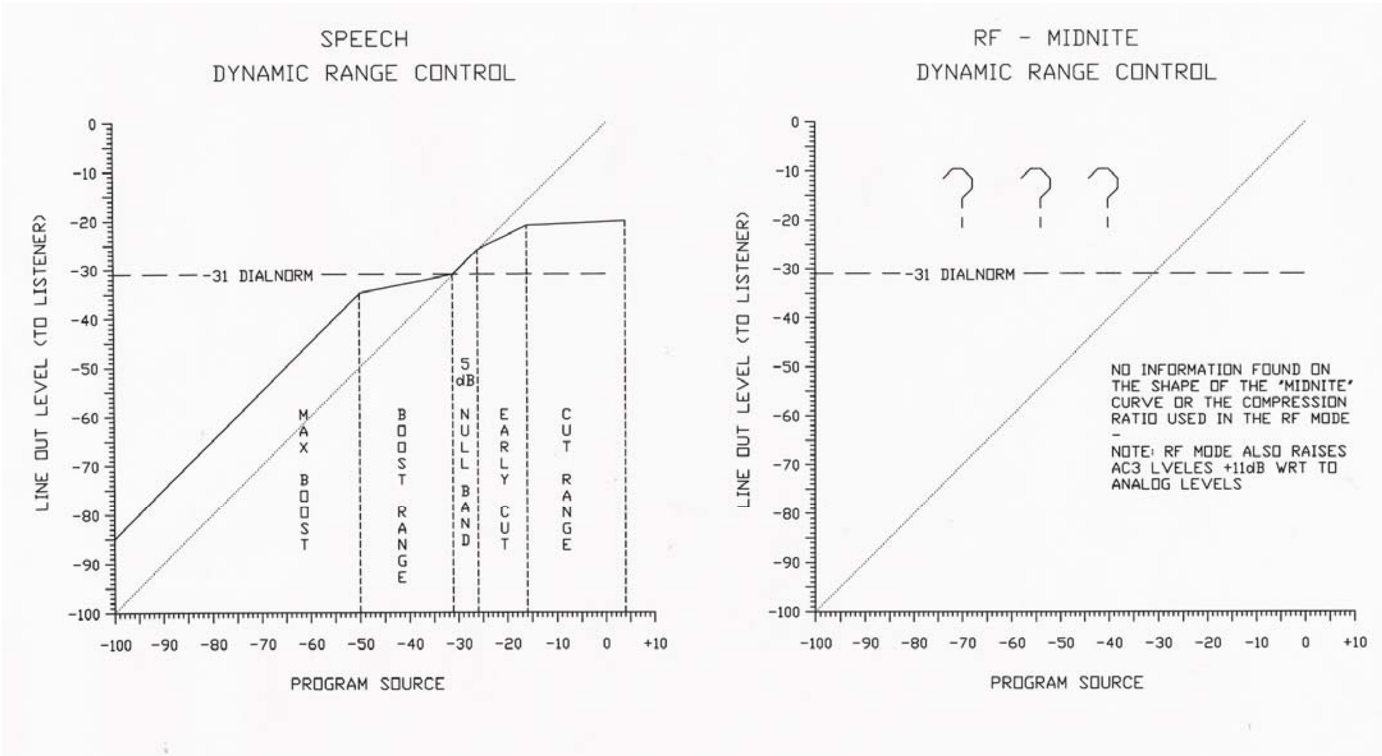


Figure 8 – DRC settings for speech and the "Midnite" mode

KEY POINTS FROM THE DRC CURVES

Note the placement of the "-31 DIALNORM" dashed line on the drawings. In the "LIGHT" DRC curves, this is centered on the 1:1 slope of the DRC curve, that is, the 20 dB range where no audio expansion or compression is occurring (called the Null Band). The Dialnorm value therefore centers on the "no processing" area of the curve. If the Dialnorm value being sent does not match the actual dialog level, that dialog will now be riding up and down on that part of the curve that will case the dialog to be unnaturally processed, resulting in "breathing" or pumping artifacts. This problem will be much worse in the "STANDARD" DRC curves, where the Null Band is only 5 dB. It would appear that you have an error of margin in setting the correct Dialnorm value of only about 2 dB.

The curves shown were drawn from data printed in the "Dolby Metadata Guide – Issue 3", available on the Dolby website. It looks like the "-31 DIALNORM' dashed line is asymmetrically placed on the "STANDARD" and "SPEECH" DRC curves, in reality, the Null Band will be centered at the value of the encoded Dialnorm value.

You should assume that most viewers will use the "STANDARD FILM" mode when watching TV, not because this is a default setting, but because is the most challenging to the source material. It has an "Early Cut" slope of 2:1 compression that the "STANDARD MUSIC" DRC does not have. In theory, the viewer's receiver will use the DRC value provided by the content provider, however, the actual DRC used may be over ridden by user preference or even the broadcasters encoder setting.

Most Home Theater Receivers seem to default to the "OFF" mode for DRC; therefore, they will allow the incoming DRC metadata to select the DRC curve.

THE COMPLETE ANSWER

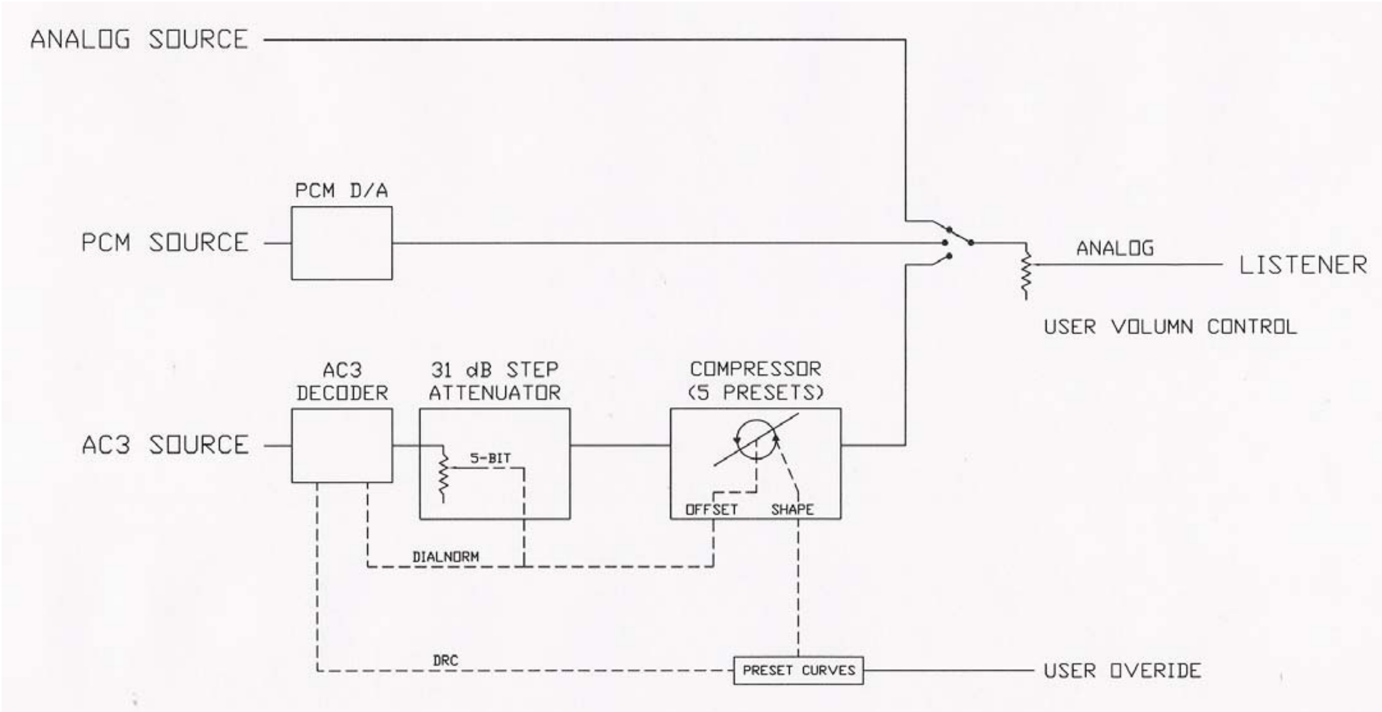


FIGURE 1A – Example of a Multi-Source Consumer Playback System

The typical home audio set-up has several types of audio to deal with. The traditional analog audio sources, vinyl records, AM and FM radio and any of several analog tape formats. There is no audio "metadata" on analog formats. Some of the analog formats use Dolby or DBX noise reduction. The playback system can not tell if and which noise reduction format was used and it is up to the end user to correctly select the proper mode, hopefully there is some label on the media to convey this information.

In the 1980's and to the present, the consumer was introduced to a string of digital audio formats, the audio CD, DAT tape, MP3, SACD and the list goes on. These are in a general class of PCM (Pulse Code Modulation). The PCM formats have some metadata carried in the bit stream but it is limited to descriptive information like bit-rate and copy protection. The PCM formats does not carry Dialnorm or DRC information.

The latest audio format presented to the consumer is Dolby Digital which raises metadata to a whole new level. The metadata not only sets up the decoder for proper decoding and copy protection, but can also set up the consumers receiving equipment, adjusting the volume and dynamic range "on the fly". For the most part the consumer can over ride the producer's artistic decisions. The consumer can also misadjust many of the menu settings now available on these systems.

The consumer is accustomed to adjusting the volume control as different analog or PCM sources are selected. For Dolby Digital sources, once the consumer sets his preferred listing level, the Dialnorm will ensure that as different Dolby Digital sources are selected (ie- changing TV channels), the volume control did not be touched again. It is our job to make that last sentence a reality. The current situation is far from the fact as my off-air measurements will demonstrate.

MULTICHANNEL DIGITAL ENCODERS

A comparison of Dolby E, AC3 and AES PCM encoders is shown on the drawing. Dolby E is a method to record up to 8 channels of audio on the AES audio input of a digital video recorder. Since the data rate of Dolby E is less than the AES channel capacity, the Dolby E data will need to be padded out with zeros. Dolby E is video frame synchronous, is not compatible with PCM decoders but can be treated as AES for distribution. Dolby E has a one-frame encoding delay and a one-frame decoding delay which Digital VTR's account for internally.

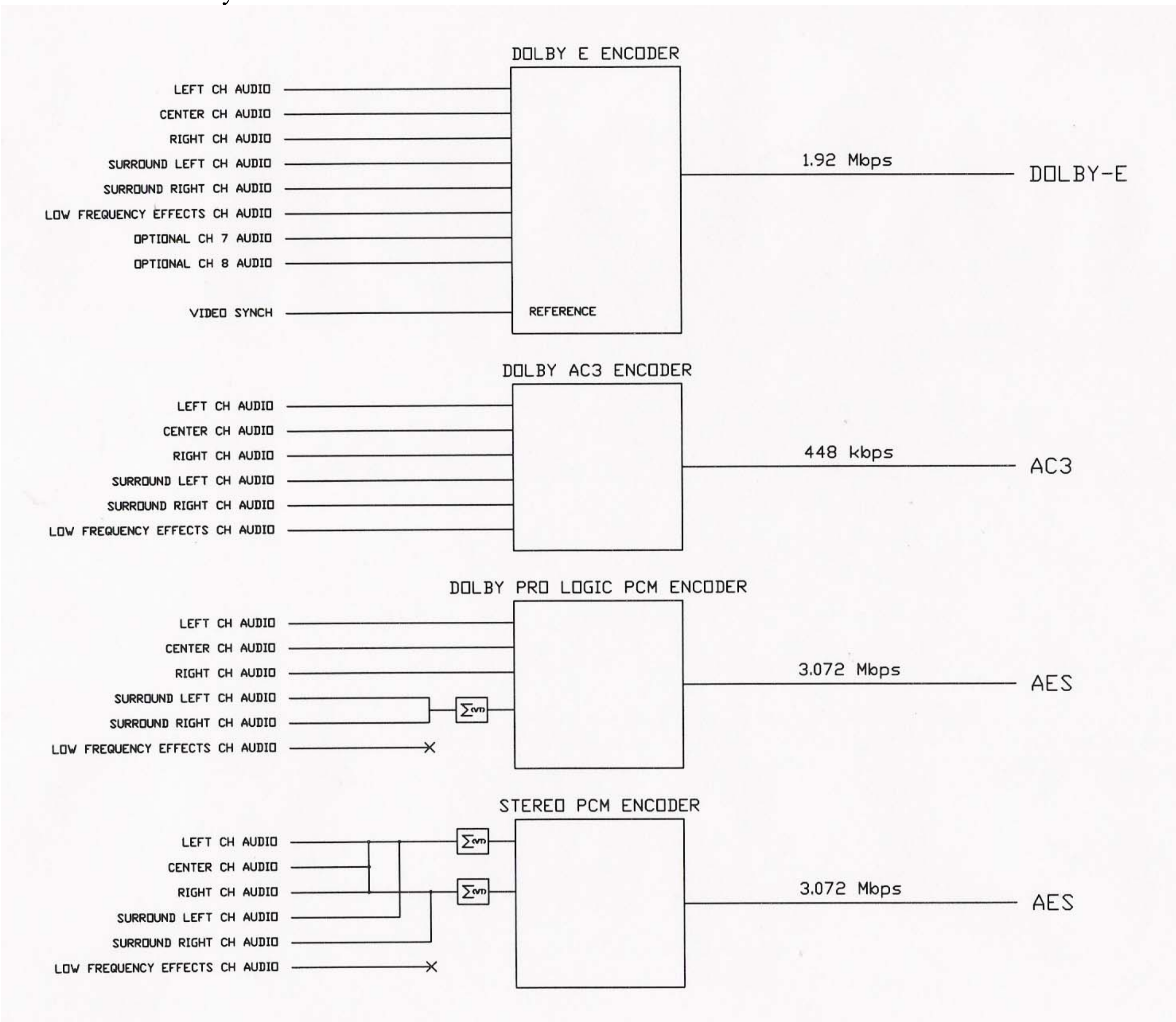


Figure 9 – Comparison of the bit-rates of various digital audio formats

The above drawing shows the encoded bit-rates of the most common professional digital audio formats. Note that the Dolby-E encoded bit-rate will fit comfortably with in the 2-channel AES (PCM) pair. This allows the 6-channel audio format to be delivered over a single AES pair. If you want to know more about Dolby-E, consult the Dolby web for more information. The key advantage of Dolby-E is that the Dolby-E frame boundaries align with the video frame boundaries. AES frame boundaries and video frame boundaries have a non-integer relationship. The "(WT)" next to the summation sign indicates a weighted sum.

CONSUMER PLAYBACK CONNECTIONS

This drawing shows how the Sub Woofer and LFE outputs of a typical high end Home Theater system are configured. Some systems delete the Sub Woofer jack, folding the L.F. content of the main channels into the LFE output.

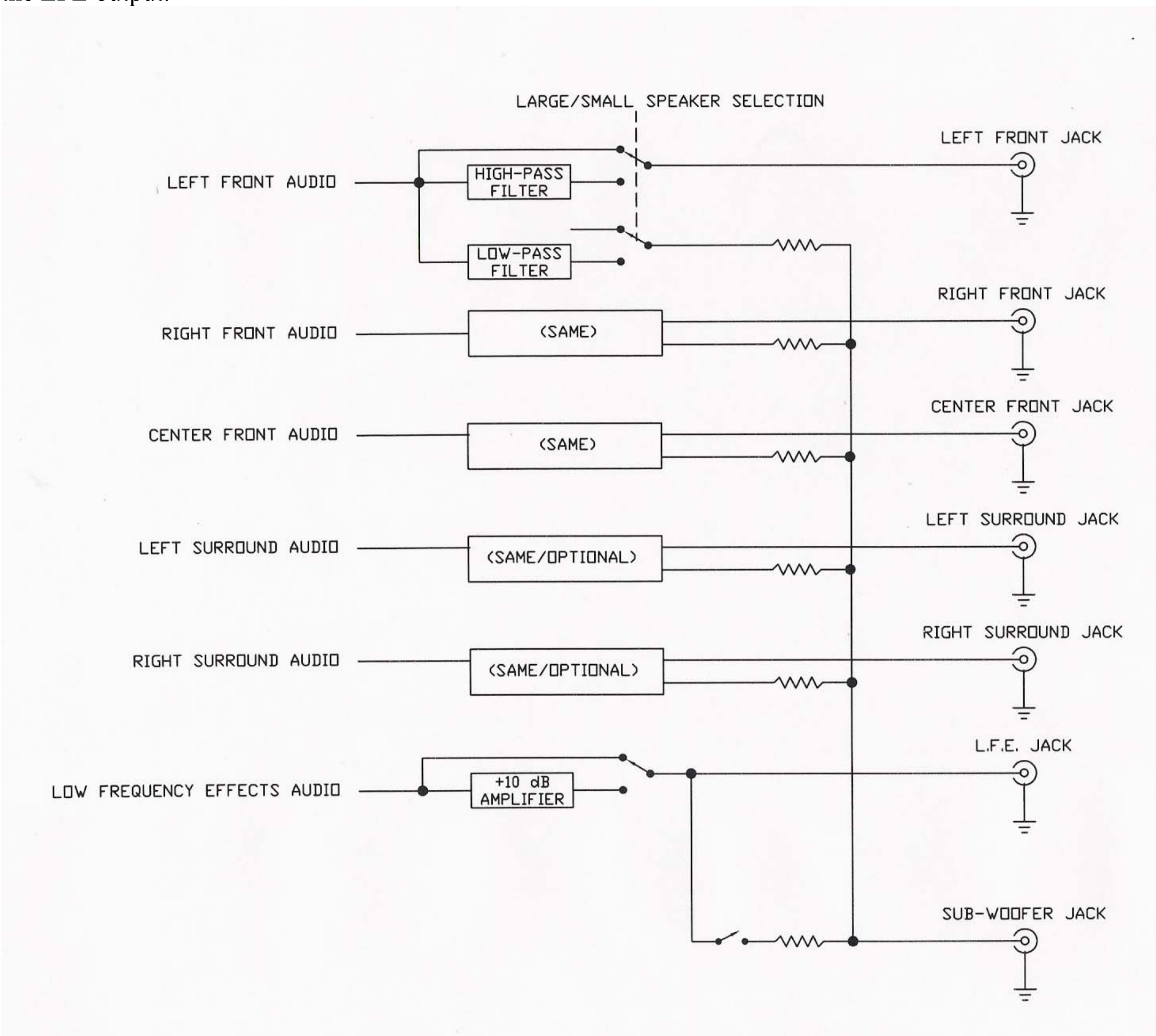


Figure 10 – Typical consumer Home Theater audio paths

Typically, there is very little signal on the LFE (Low Frequency Effects) channel. If you refer back to Figure 9, you will note that when 6-channel audio is encoded into stereo PCM or Dolby Surround, the LFE channel information is discarded! The LFE channel bandwidth is roughly 20-120 Hz, if your newscast has a lot of signal in the LFE channel, something is not right. What the LFE speaker (the sub-woofer) is primarily used for in the home theater system is to reproduce the bass content the typical small front and center channel speakers are incapable of handling. Note that the LFE channel applies 10 dB of gain to the signal feeding the sub-woofer. This gain stage will be either in the receiver circuitry or sometimes within a powered sub-woofer. This playback 10 dB gain stage is part of the Dolby Digital standard, and if not attended to properly may lead to a 20 dB error in the sub-woofer levels. Some receivers do not have separate LFE and SUB-WOOFER jacks.

DOLBY LM100

The Dolby LM100 (MSRP $3,200 JAN 2010 pricing) is the currently available method of measuring the actual Dialnorm value and reading the Dialnorm value present in Dolby Digital streams. It can measure analog, PCM, Dolby-E and AC3 (Dolby Digital) audio sources and calculate a suggested Dialnorm value for insertion in the metadata. A number of manufactures are stating to ship audio meters capable of measuring dialog loudness levels using the Leq (A) & LKFS algorithms. Expect to see more appear on the market once the C.A.L.M Act becomes law

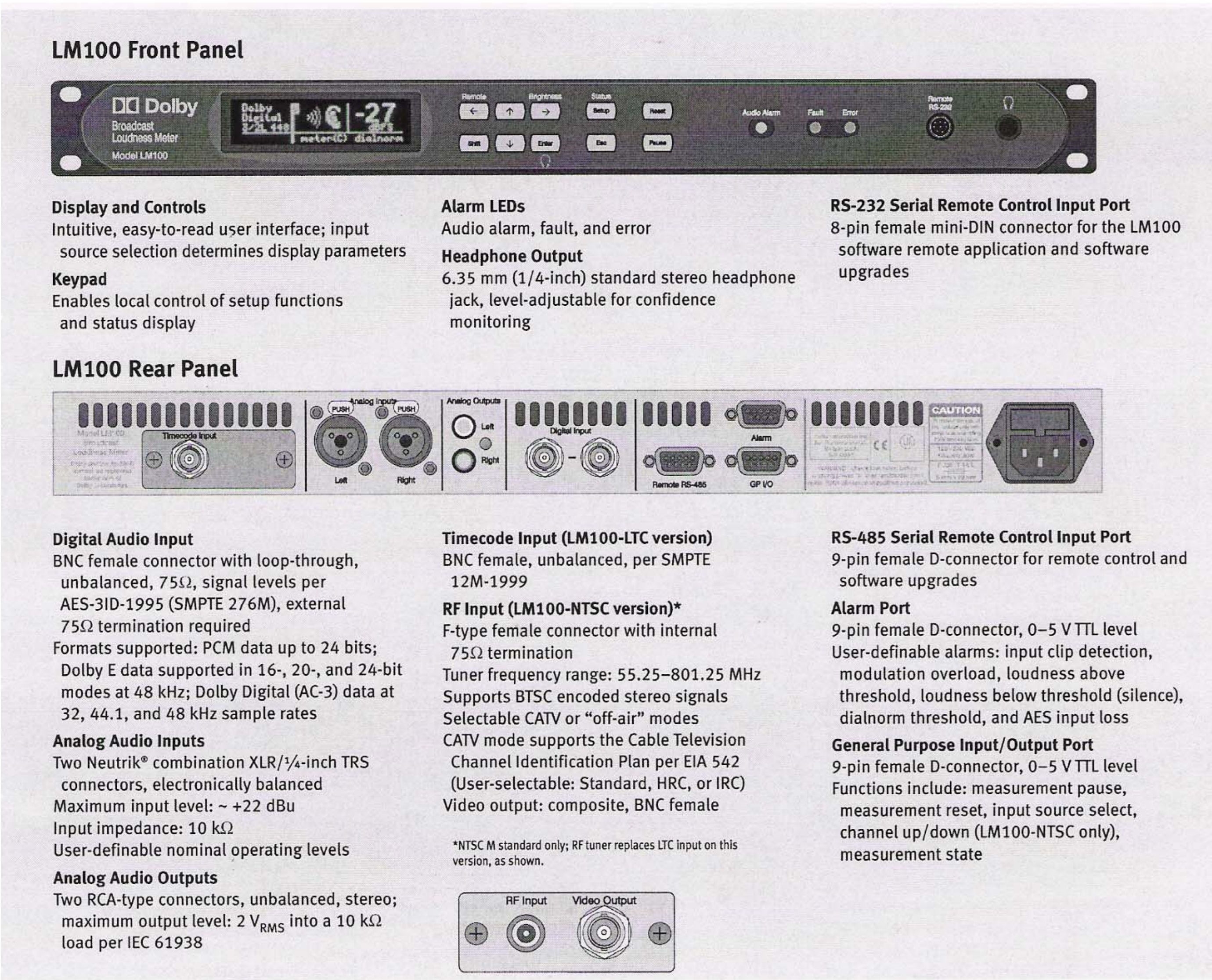


Figure 11 – The Dolby LM100

The LM100 was designed some time ago. The connection to your computer for logging data is via either a front panel RS-232 connection which requires a special cable (provided with the LM100), or a rear panel RS-422 DB9F connector, sorry, no USB or LAN connectivity. The remote control and data collection software is supplied with the LM100. The digital input, for use with either AES (PCM), Dolby Digital and Dolby-E signals in via a coaxial BNC connector. To analyze optical Dolby Digital (AC3) consumer audio formats, you'll need an optical to coax converter. I have used the HOSA model ODL-276A (about $80) successfully.

OBSERVATIONS ON THE LM100

The LM100 is purpose built to do two things. Read the Dialnorm data in Dolby Digital audio streams and to determine the Dialnorm value of analog, AES digital and Dolby Digital audio programs for insertion in AC3 metadata. The LM100 is similar in operation to the VU meter that you are accustomed to; however; a VU meter, PPM meter or “Loudness” meter can not be used to determine an accurate Dialnorm value. The LM100 has a special algorithm to measure the subjective loudness based on one of two standards. The original standard was called Leq(A), which was called the loudness equivalency, A-weighted. The newer, more preferred method is called the ITU-R BS.1770 standard, which stands for Loudness, K-weighting, Full Scale. There is only about a 2 dB difference between to two standards, but as I mentioned earlier, a 2 dB error can be significant. The Dialnorm measurement is taken only during the dialog portions of the program using a Dialog Intelligence algorithm. The Dialog Intelligence can be turned off for determining the Dialnorm value of music or non-dialog program material.

If you have an older LM100 without the new ITU-R standard, you can upgrade your unit with a simple $10 firmware upgrade that can be ordered from your local Dolby re-seller.

Ideally, the most accurate Dialnorm measurement is taken by having the LM100 set to take a long term average over the duration of the program in question. The LM100 will also take short term measurements to allow and audio operator to compare how the audio console metering compares with the LM100 readings. As a very unscientific test, I fed some audio test tones into the LM100 to see what kind of Dialnorm readings I would obtain with Dialog Intelligence on and off. I used AES and analog signals.

Signal	Level	Frequency	Dialog Intelligence ON Short Term Dialnorm	Dialog Intelligence OFF Short Term Dialnorm
PCM	-20 dBFS	400 Hz	-22	-22
		1000 Hz	-17	-17
		2000 Hz	-16	-16
		4000 Hz	-16	-16
		8000 Hz	-19	-16
PCM	-30 dBFS	400 Hz	-32	-31
		1000 Hz	-28	-27
		2000 Hz	-26	-26
		4000 Hz	-26	-27
		8000 Hz	-29	-29
ANALOG	0 dB	400 Hz	+2	+2
		1000 Hz	+3	+4
		2000 Hz	+2	+4
		4000 Hz	+4	+2
		8000 Hz	+3	+1
ANALOG	-10 dB	400 Hz	-10	-12
		1000 Hz	-6	-9
		2000 Hz	-6	-7
		4000 Hz	-8	-6
		8000 Hz	-9	-7

Table 2 - I have no further comment on the above readings except, hmmmm, interesting.

From my very crude analysis, it is evident that the LM100 is not a traditional VU meter. Since DialNorm is a time averaged reading, the DialNorm meter is not a substitute for a traditional VU meter and requires some experimenting to obtain accurate DialNorm readings.

LM100 APPLICATION EXAMPLE

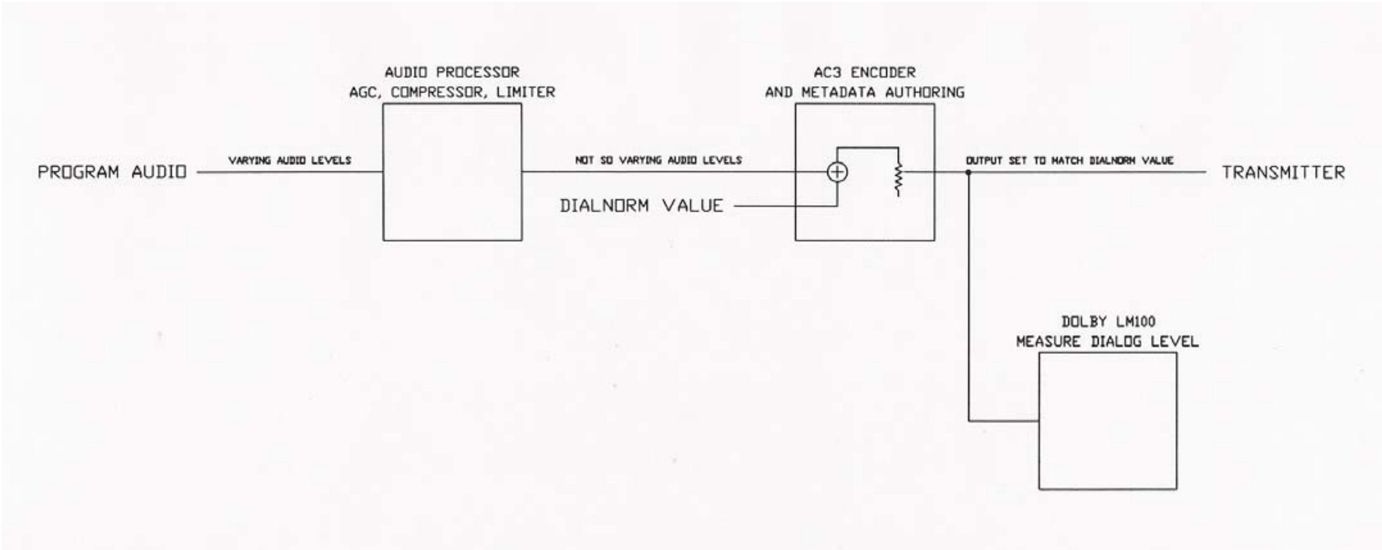


Figure 11A – Using the LM100 to calibrate the Dialnorm setting

When digital TV was first being broadcast, it was assumed that since digital audio had practically no noise floor issues, a near 100 dB SNR and extremely low distortion, the days of audio processing were over. We soon learned that just going digital did not solve the legacy analog problems of level variations and over and under "modulation". Since we were simulcasting on both our analog NTSC and digital ATSC transmitters, level matching between the two transmission modes became a priority.

Digital audio processors capable of handling 6-channels of audio were soon introduced to provide AGC, compression and limiting functions, similar to analog processors, but not as aggressive.

Most AC3 encoders have a data port to accept real time Dialnorm metadata. Since the reality is that dynamic metadata synchronous with the audio is not yet a reality, most broadcasters used a fixed value of Dialnorm. The output level of the AC3 encoder is then adjusted until the LM100 indicates a measured Dialnorm equal to the fixed Dialnorm value programmed into the encoder.

There is currently under discussion is what the "standard" Dialnorm value should be. Originally, the standard Dialnorm value was -27, apparently based on the default settings of Dolby Digital encoders used by program content producers. The ATSC Recommended Practice A/85 now recommends a default Dialnorm value of -24 LKFS (dialog level based on the newer ITU-R BS.1770 recommendation).

The actual Dialnorm value used is not as important as ensuring that the transmitted dialog level matches the transmitted Dialnorm value. If one station is set to -24 and another is set to -27, at the consumer end, both stations will sound equally loud (refer back to Figures 1 and 1A)

OFF-AIR MEASUREMENTS

The balance of the presentation is a number of off-air measurements taken of the local Madison market TV stations. Some of the issues noted may have been addressed by the time this is published. This is followed by a live demonstration of the LM100 using off-air signals and DVD material.

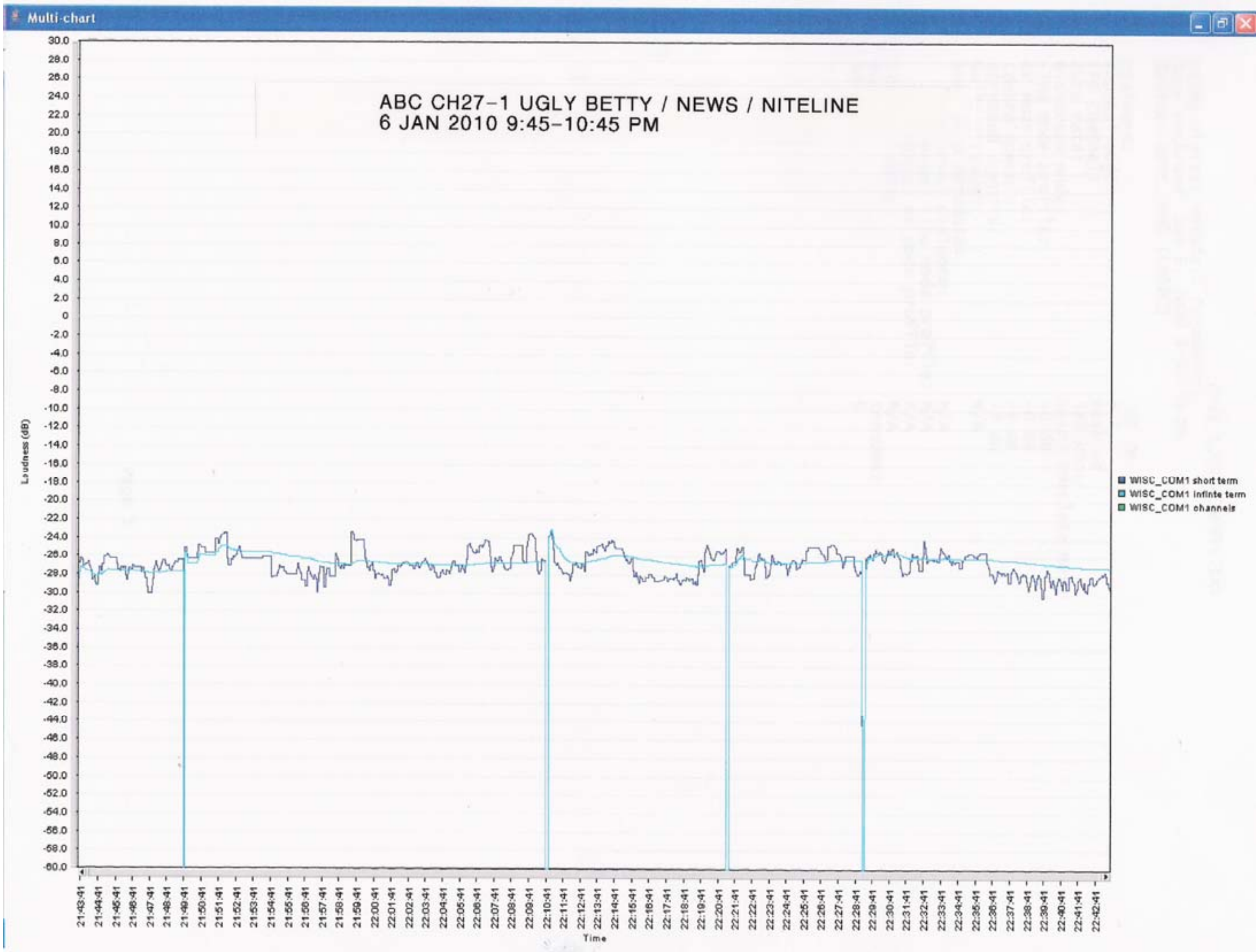


Figure 12 –

UGLY BETTY PARAMETERS		10 PM NEWS PARAMETERS	
Dialnorm:	-27 dB	Dialnorm:	-27 dB
Channel mode:	3/2	Channel mode:	3/2
LFE Channel:	Enabled	LFE Channel:	Enabled
Data rate:	448 kbps	Data rate:	448 kbps
Bitstream mode:	Main: Complete Main	Bitstream mode:	Main: Complete Main
Line mode profile:	-0 dB	Line mode profile:	0 dB
RF mode profile:	-0 dB	RF mode profile:	1 dB
Centre downmix:	-3 dB	Centre downmix:	-3 dB
Surround downmix:	-3 dB	Surround downmix:	-3 dB

NOTES- The audio hits (4) were of unknown origin and did not coincide with commercial breaks.

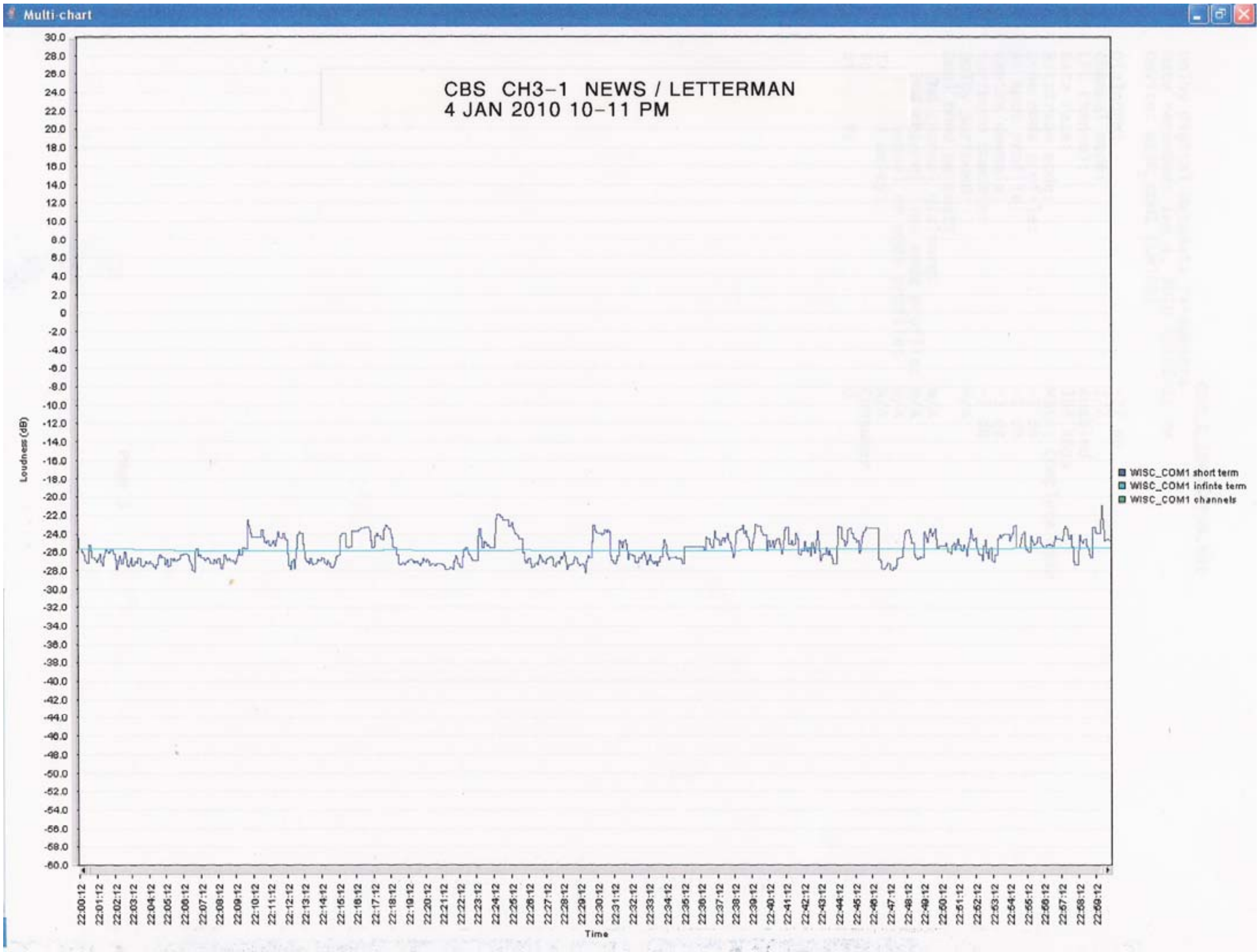


Figure 13 –

10 PM NEWS PARAMETERS

Dialnorm:	-27 dB
Channel mode:	3/2
LFE Channel:	Enabled
Data rate:	384 kbps
Bitstream mode:	Main: Complete Main
Line mode profile:	-1 dB
RF mode profile:	-1 dB
Centre downmix:	-3 dB
Surround downmix:	-3 dB

LETTERMAN PARAMETERS

Dialnorm:	-27 dB
Channel mode:	3/2
LFE Channel:	Enabled
Data rate:	384 kbps
Bitstream mode:	Main: Complete Main
Line mode profile:	-2 dB
RF mode profile:	-3 dB
Centre downmix:	-3 dB
Surround downmix:	-3 dB

NOTES- Measured Dialnorm 1 dB hotter than Dialnorm Metadata

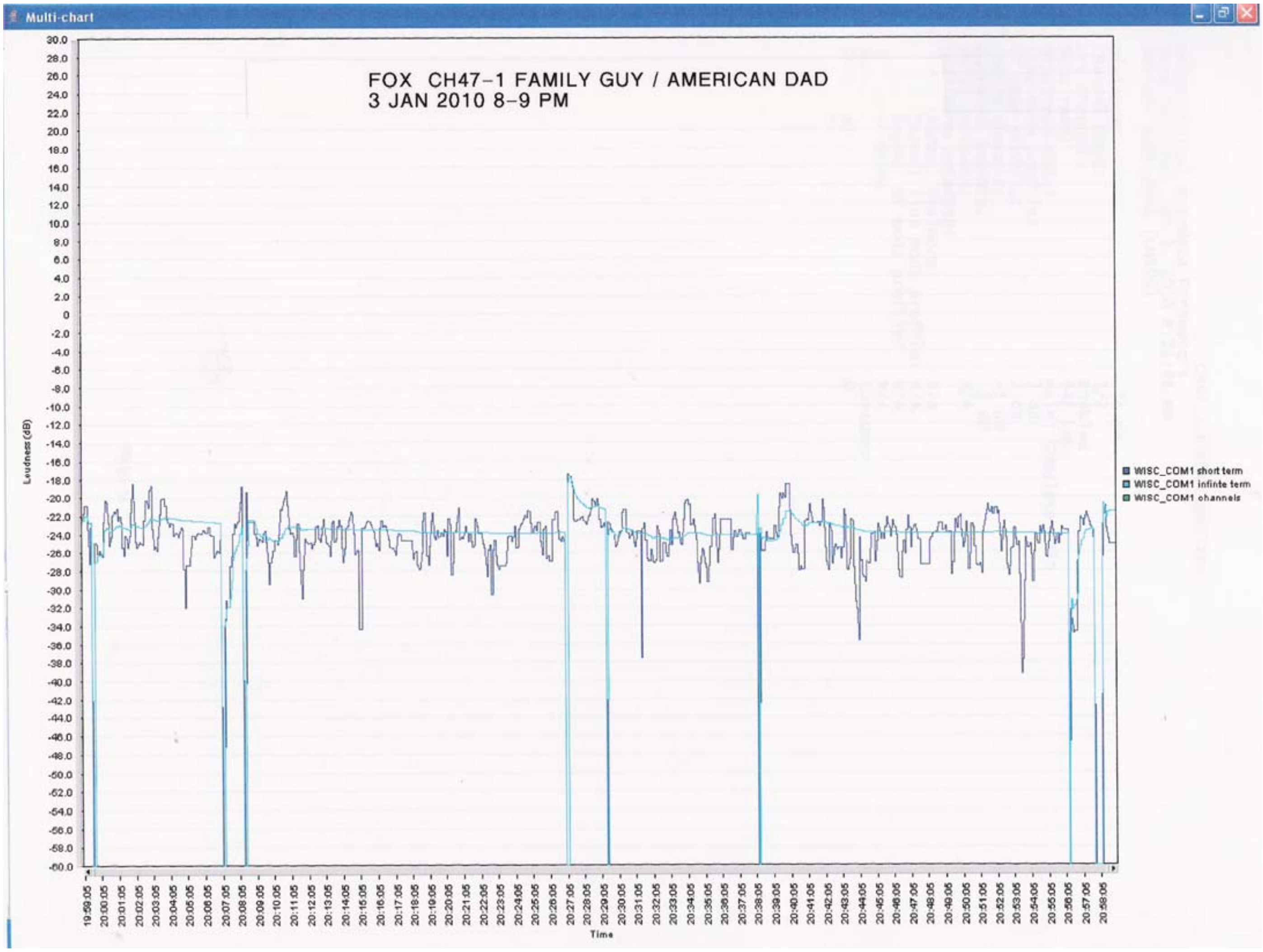


Figure 14 –

FAMILY GUY PARAMETERS

Dialnorm:	-24 dB
Channel mode:	3/2
LFE Channel:	Enabled
Data rate:	448 kbps
Bitstream mode:	Main: Complete Main
Line mode profile:	2 dB
RF mode profile:	2 dB
Centre downmix:	-3 dB
Surround downmix:	-3 dB

AMERICAN DAD PARAMETERS

Dialnorm:	-24 dB
Channel mode:	3/2
LFE Channel:	Enabled
Data rate:	448 kbps
Bitstream mode:	Main: Complete Main
Line mode profile:	-2 dB
RF mode profile:	-3 dB
Centre downmix:	-3 dB
Surround downmix:	-3 dB

NOTES- Data hits occur at each local break causing audible low frequency sound. Suspect it may be related to 310M bit-splicing being used.

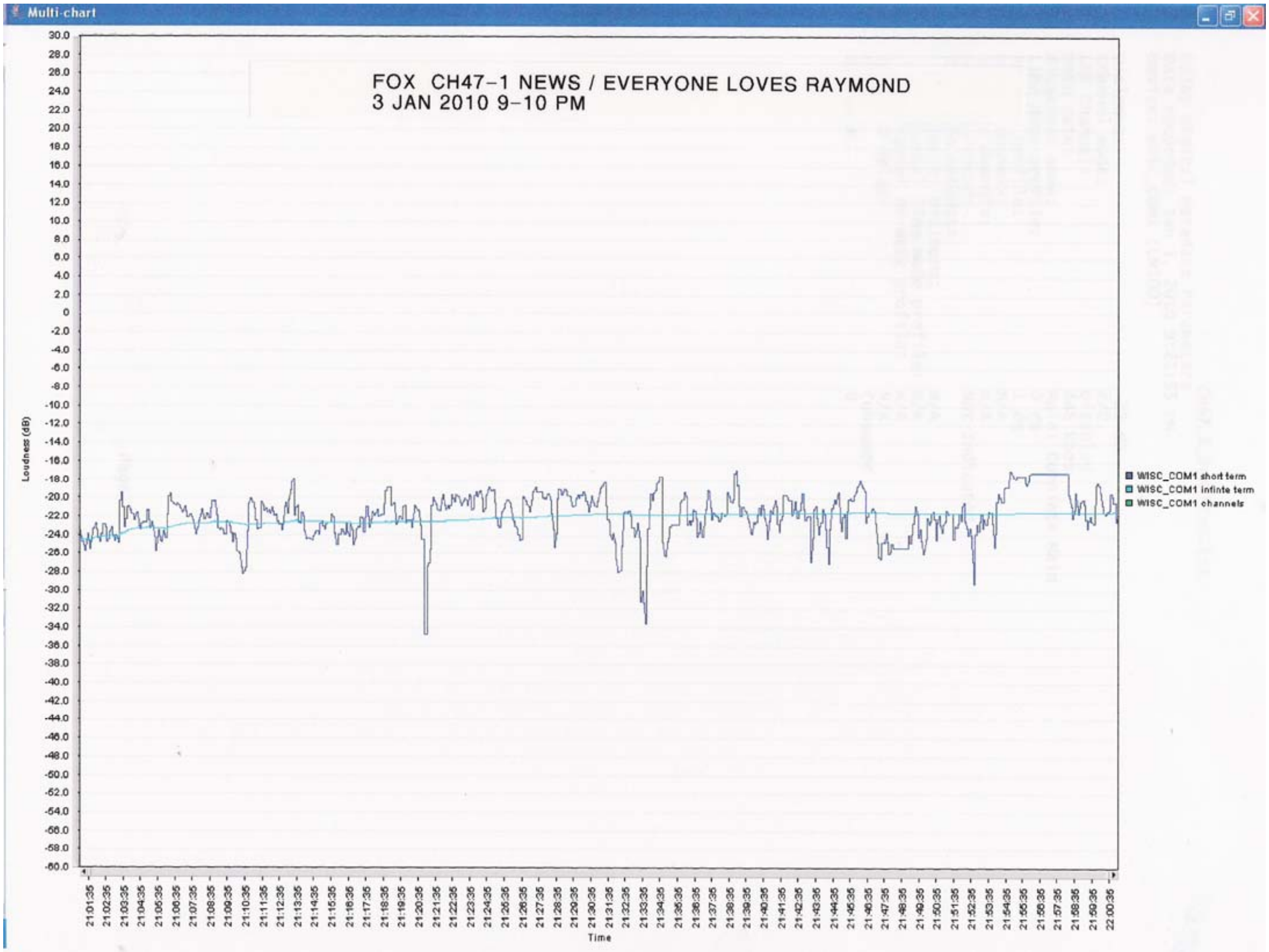


Figure 15 –

9 PM NEWS PARAMETERS	
Dialnorm:	-23 dB
Channel mode:	2/0
LFE Channel:	Disabled
Data rate:	448 kbps
Bitstream mode:	Main: Complete Main
Line mode profile:	0 dB
RF mode profile:	1 dB
Centre downmix:	N/A
Surround downmix:	N/A

RAYMOND PARAMETERS	
Dialnorm:	-23 dB
Channel mode:	2/0
LFE Channel:	Disabled
Data rate:	448 kbps
Bitstream mode:	Main: Complete Main
Line mode profile:	1 dB
RF mode profile:	1 dB
Centre downmix:	N/A
Surround downmix:	N/A

NOTES- Once in local programming, data hits stop.

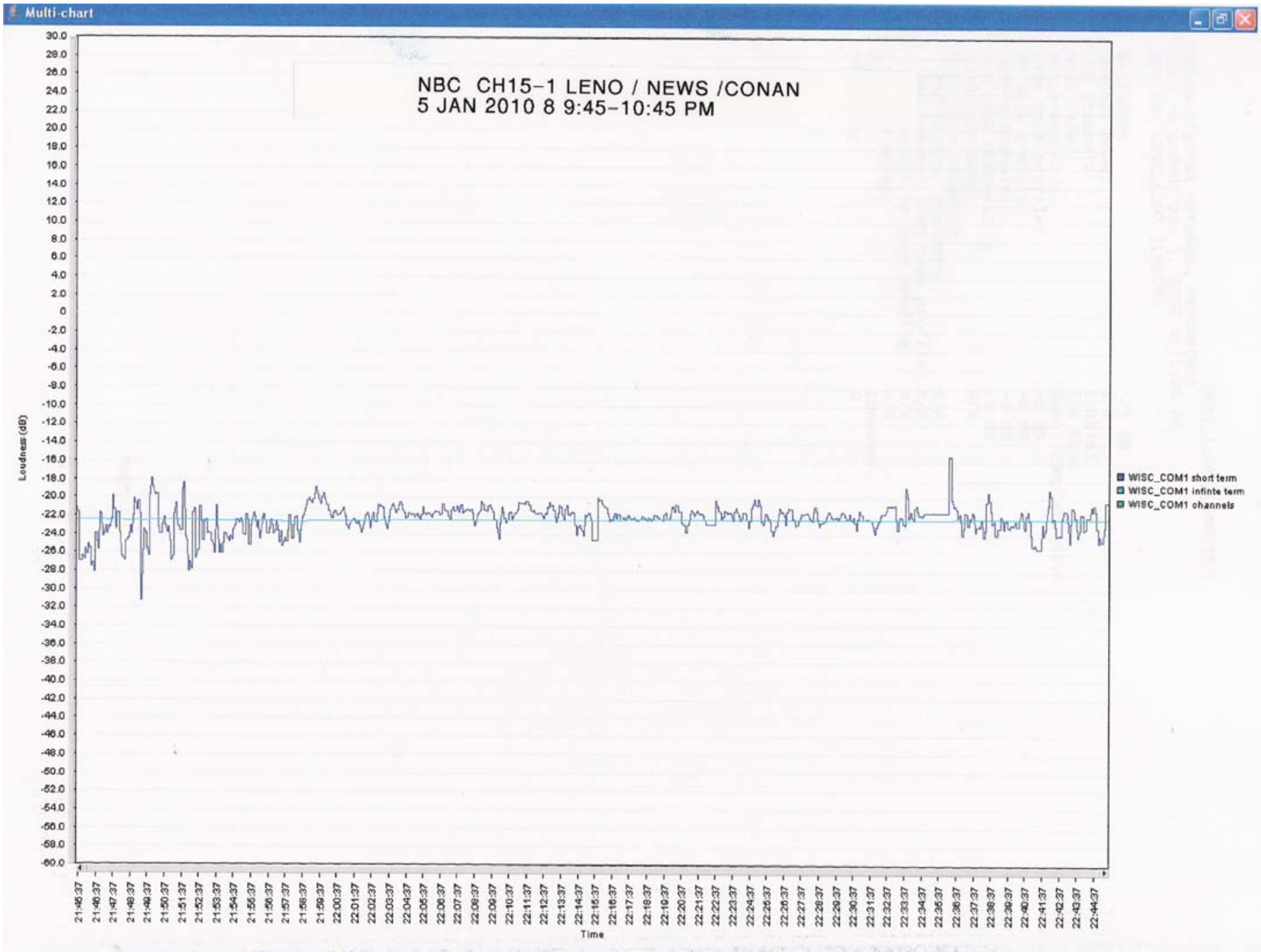


Figure 16 –

10 PM NEWS PARAMETERS		LENO PARAMETERS	
Dialnorm:	sorry-forgot to take	Dialnorm:	-24 dB
Channel mode:	sorry-forgot to take	Channel mode:	3/2
LFE Channel:	sorry-forgot to take	LFE Channel:	Enabled
Data rate:	sorry-forgot to take	Data rate:	384 kbps
Bitstream mode:	sorry-forgot to take	Bitstream mode:	Main: Complete Main
Line mode profile:	sorry-forgot to take	Line mode profile:	-0 dB
RF mode profile:	sorry-forgot to take	RF mode profile:	-0 dB
Centre downmix:	sorry-forgot to take	Centre downmix:	-3 dB
Surround downmix:	sorry-forgot to take	Surround downmix:	-3 dB

NOTES- News block appears more heavily compressed that network progrmming

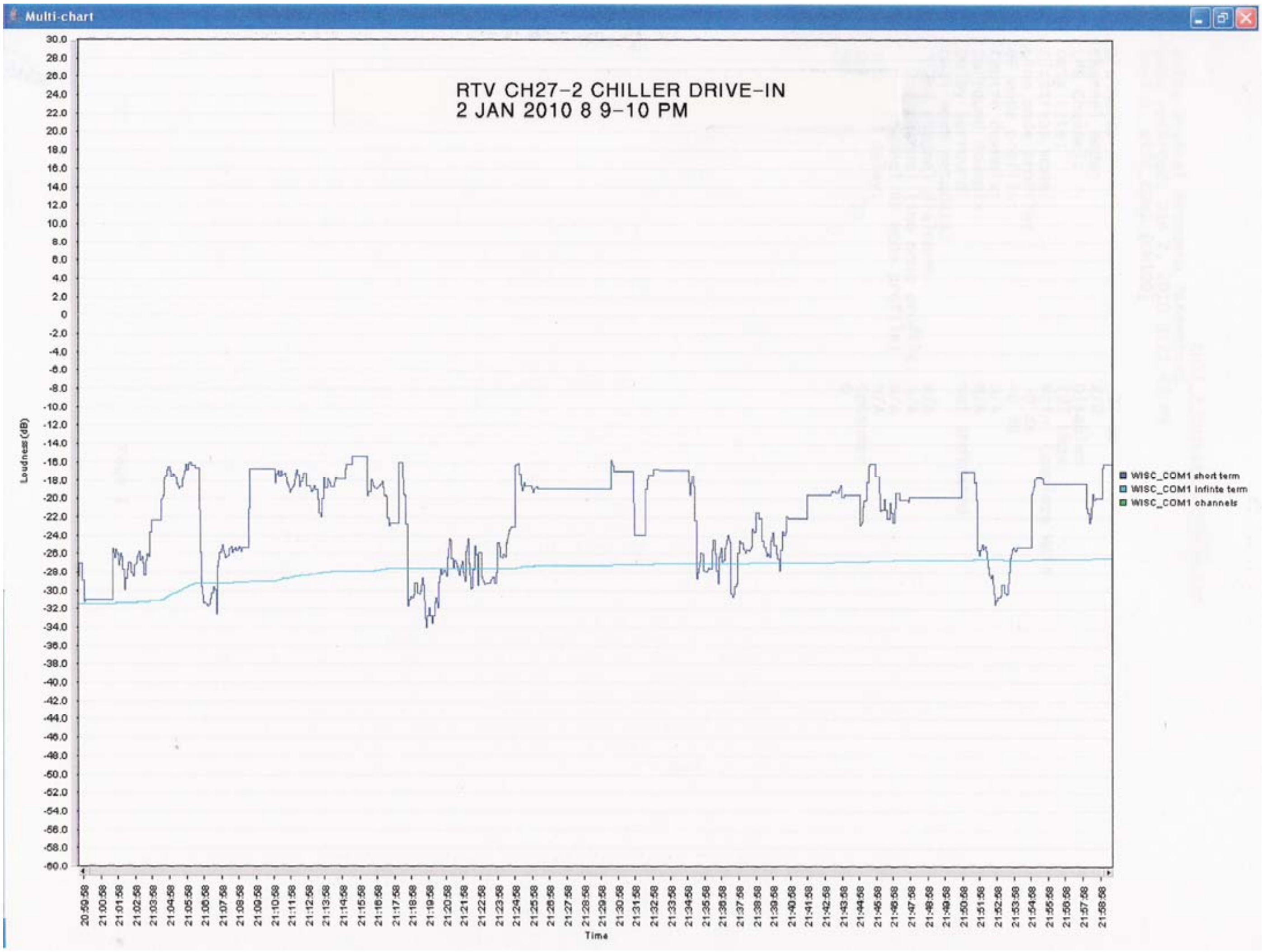


Figure 17 –

CHILLER DRIVE IN PARAMETERS

Dialnorm:	-27 dB
Channel mode:	2/0
LFE Channel:	Disabled
Data rate:	192 kbps
Bitstream mode:	Main: Complete Main
Line mode profile:	-0 dB
RF mode profile:	-0 dB
Centre downmix:	N/A
Surround downmix:	N/A

NOTES- Program audio is 15 dB hotter than commercials.

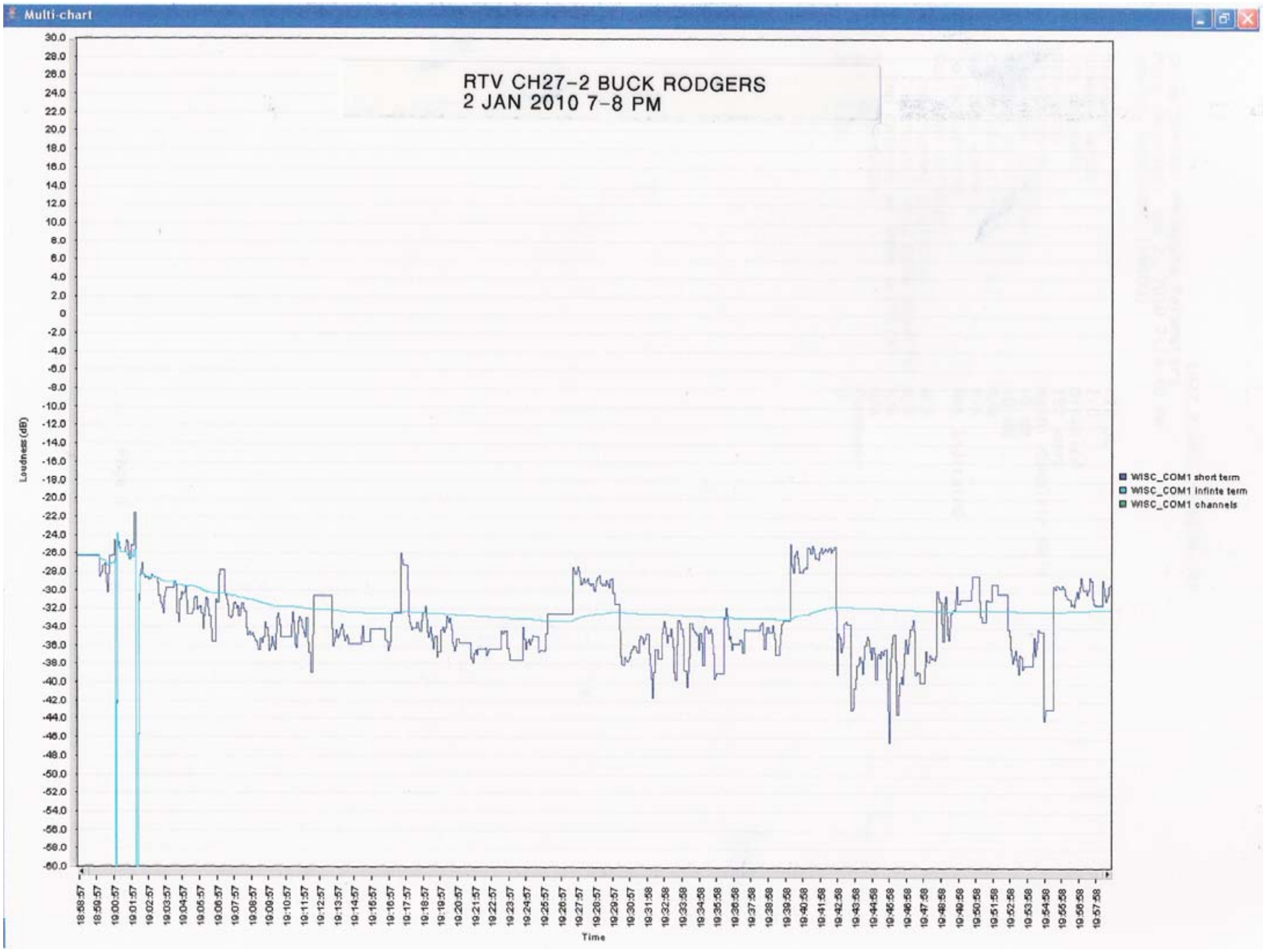


Figure 18 –

BUCK RODGERS PARAMETERS

Dialnorm:	-27 dB
Channel mode:	2/0
LFE Channel:	Disabled
Data rate:	192 kbps
Bitstream mode:	Main: Complete Main
Line mode profile:	-0 dB
RF mode profile:	-0 dB
Centre downmix:	N/A
Surround downmix:	N/A

NOTES- Commercial audio is 10 dB hotter than program.

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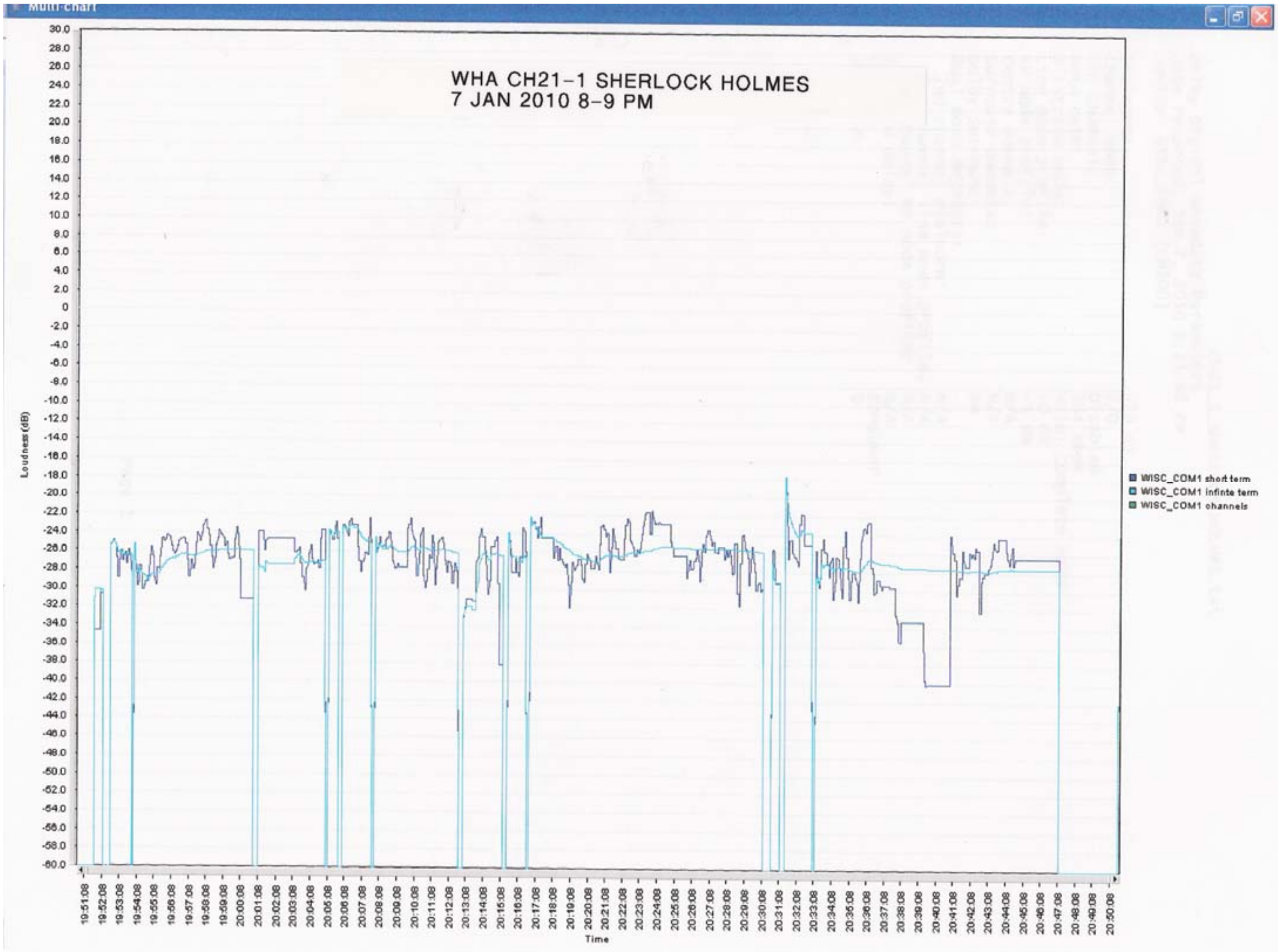

Figure 19 –

SHERLOCK HOMES PARAMETERS

Dialnorm:	-28 dB
Channel mode:	2/0
LFE Channel:	Disabled
Data rate:	384 kbps
Bitstream mode:	Main: Complete Main
Line mode profile:	-0 dB
RF mode profile:	-1 dB
Centre downmix:	N/A
Surround downmix:	N/A

NOTES- Multiple audio hits of unknown cause resulting in audio mutes.

REFERENCES

ATSC standards	www.atsc.org	
Dolby Products	www.dolby.com	
Dolby Support	www.dolbysupport.com	need to register, need valid Dolby product s/n
JBL Professional	www.jblpro.com	
Linear Acoustic	www.linearacoustic.com	
Test media	www.audioxpress.com	

The author welcomes your comments, suggestions, improvements and corrections to the material presented.

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