Dolby Laboratories Recommended Practices For Cable Television Systems

Measuring Equivalent Loudness of Speech in Analog & Digital Audio

DRAFT[1]

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[1] This document is classified as a "work in progress" and is continuously being updated.

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1. Scope

The scope of this recommended practice is to provide guidance with the measurement and provisioning of analog (NTSC) audio modulation, digital simulcast, DPI, and VOD encoding equipment in an effort to more consistently match the reproduced levels of speech between analog and digital program services.

This document also provides assistance to cable television technical personnel in understanding the application of the Dolby Laboratories LM100-NTSC Broadcast Loudness Meter with Dialogue Intelligence™ in these applications.

2. Informative References

47 C.F.R. §73.682

ATSC A/52B, Digital Audio Compression (AC-3), Revision B

CEA CEB11, NTSC / ATSC Loudness Matching

IEC 61672-1, Electroacoustics – Sound Level Meters – Part 1: Specifications

OpenCable™ Host Device Core Functional Requirements, OC-SP-HOST-CFR-I15-031121

NCTA Recommended Practice for Measurement on Cable Television Systems, 3rd Edition

3. Discussion

Digital set-top boxes and the technologies contained within them have considered, during their design, the ability to reproduce audio levels (on both analog and digital program services) that subjectively match each other. Cable customers tend to set their listening level based on the acoustic level of speech contained within the selected program. Speech is an essential component of virtually all television programming. Therefore, it is desired that digital and analog programming deliver consistent speech levels in an effort to keep the subscriber from having to adjust the level control significantly and frequently. CEA document CEB11 and CableLabs OpenCable™ host device core functional requirements document OC-SP-HOST-CFR-I15-031121 provide guidance to set-top box manufacturers concerning the audio gain structure within the set-top in order to provide subjective loudness matching between digital and analog program sources.

The equivalent loudness of speech (A-weighted) for analog audio sources is typically 17dB below 100% modulation for monophonic programming or for both channels combined (as the power sum) for BTSC stereo services. **Therefore it is extremely important to provision analog audio modulation equipment so that the A-weighted average speech level is ~ 17dB below 100% modulation, hereinafter referred to as –17dBFM Leq(A)[2].**

Figure 1 shows the level relationships between the output of the digital program audio decoder and analog program audio sources on the modulated output (i.e. RF) of a typical digital set-top box. Notice that the NTSC scale has a maximum value of 0dBFM and that this level is equivalent to 100% modulation as per FCC rules (i.e. 25kHz peak deviation)[3] and the Leq(A) speech level is shown at 17dB below 100% modulation. This value (-17 dBFM) indicates the ratio between the maximum program peaks and the average Leq(A) speech level.

Every digital set-top box includes a 2-channel Dolby Digital (AC-3) decoder that can be operated in two modes, Line and RF[4]. Figure 1 shows that the maximum permissible level of +6 dBFM which is equivalent to 200% modulation (i.e. 50kHz peak deviation) for digital program audio is available at this output. This level relationship is intentional since the digital program audio being decoded by the set-top potentially has at least, 6 dB more headroom above speech peaks than analog program audio (i.e. while the decoder is operating in RF mode). Furthermore, since the BTSC stereo and SAP system leads to a peak deviation of 73kHz most television tuners today can accept up to 8dB above 25kHz peak deviation in the absence of pilot and subcarriers without distortion[5].

Figure 1 shows that with the set top decoder operating in RF mode the decoded speech level for digital program audio will match the analog programming speech level. More importantly, note that if the set-top decoder is operating in Line mode that the decoded level of speech in the digital program audio is reproduced at –28 dBFM Leq(A) or 11 dB lower than the analog program audio. Hence, when considering the reproduced level of speech, in order for analog program audio to match digital program audio via the RF interface, the digital decoder must be operating in RF mode[6] and cable systems must also provision their analog modulation equipment so that the speech levels are at ~ -17dBFM Leq(A) as depicted in Figure 1.

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[2] Leq(A) and "Equivalent Loudness" is a common abbreviation / nomenclature for measurement devices that comply with IEC 61672-1.

[3] Many RF modulators contained in digital set-top boxes support only a single channel of audio (i.e. monophonic).

[4] Line and RF modes are described in Section 4.

[5] The FCC limits are of no significance to RF modulators in set-top boxes and VCRs.

[6] Many set-top boxes and the software applications that reside in them may utilize different nomenclature for RF and Line mode operation. Please refer to Table 1 for a comprehensive listing.

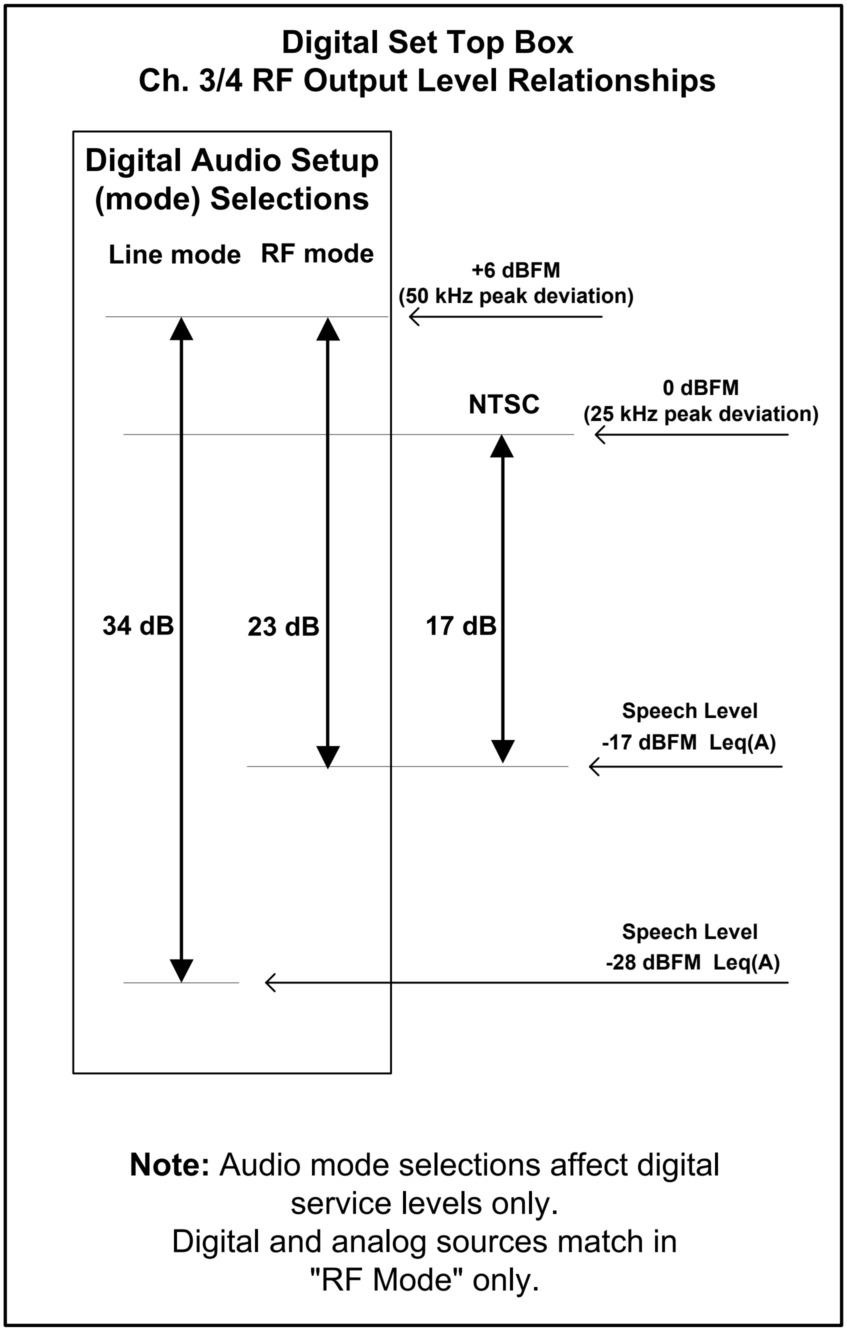

Figure 1: Analog adjusted so that the level of speech is ~ -17 dBFM Leq(A)[7]

[7] i.e., below 100% modulation 25kHz peak deviation.

4. Digital Set-top Audio Decoder Modes & Nomenclature

Dolby Digital (AC-3) decoders found in digital set-tops, in general, can operate in two modes, Line and RF. Each of these modes has a specific application, and care **must** be taken when the set top is both designed and deployed to ensure that the intended mode is used by default. This section describes the differences between, and the applications of, Line and RF mode operation within a 2-channel Dolby Digital (AC-3) decoder. Also note, differences exist among digital set top manufacturers and guide application providers as to the nomenclature used to indicate Line and RF mode operation to the user. Table 1 shows the equivalent modes for the most common digital set tops used today.

Line Mode Operation:
Line Mode operation was designed for, and generally applies to the baseband line level outputs of two-channel set-top decoders, two-channel digital televisions and multichannel home theatre decoders. It is important to note that ***access*** to Line Mode operation is a requirement (albeit, it should not be the default operating mode, especially if the digital set top has a channel ¾ remodulated output) for all digital cable set top boxes that have analog baseband (i.e. line level) outputs.

The decoder's outputs operating in this mode will typically be connected to a much higher quality sound reproduction system than that found in a typical television set. In this mode, dialogue normalization[8] is enabled and applied in the decoder <u>at all times</u>. In this mode the normalized level of dialogue is reproduced at a level of -31 dBFS Leq(A), but **ONLY** when the transmitted *dialnorm* value has been correctly adjusted for a particular program (See Figure 1)[9]. In general, with the reproduced dialogue level at –31dBFS, this mode allows wide dynamic range programming to be reproduced without any peak limiting and/or compression applied as may be intended by the original program producers. Furthermore, since the Dolby Digital (AC-3) digital audio coding system can provide more than 100dB of dynamic range, there is no technical reason to encode the dialogue at or near 100% as is commonly practiced in analog television systems. This allows the delivery system to meet one of its goals of being able to deliver high impact cinema type sound to the digital subscriber's living room.

RF Mode Operation:
RF Mode is intended for products such as digital cable set-top terminals that generate a monophonic signal for transmission via the channel 3/4 remodulator that feeds the RF (antenna) input of a television set. This mode was specifically designed to match the average reproduced dialogue level and dynamic range of digital sources to those of existing analog sources such as NTSC and analog cable TV broadcasts. In this operating mode dialogue normalization is enabled and applied in the decoder <u>at all times</u>. However, the dialogue level in this mode is reproduced at a level of -20 dBFS Leq(A) **ONLY** when the transmitted *dialnorm* value is valid for a particular program (See Figure 1). Thus, the Dolby Digital decoder introduces an +11 dB gain shift and therefore the maximum possible peak to dialogue level ratio is reduced by 11dB,

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[8] Refer to ATSC Document A/52a

[9] The *dialnorm* value is set by the programmer.

when compared to Line Mode. **Thus, it is important that digital set top boxes which include an RF remodulator are required to default to RF Mode as per this recommendation.**

Table 1 shows the differences in nomenclature (for Line and RF digital audio decoder operation) utilized by the most common digital set top manufacturers and guide application providers (if applicable)[10].

This recommendation strongly advises that all digital set top decoders be defaulted to RF mode operation as per Table 1. As an example, systems utilizing Scientific-Atlanta set top boxes must be defaulted to "Narrow" as this mode is equivalent to RF mode.

Failure to comply with these recommendations will most likely yield significant differences in the reproduced levels between analog and digital program sources. This behavior is clearly depicted in Figure 1 where the decoded digital speech level in RF mode matches NTSC sources, that were, or assumed to be adjusted (by a headend technician or engineer) to average -17dBFM Leq(A) as per Section 6 of this document. Whereas, decoded digital speech levels in Line mode do not match NTSC sources. (i.e. they are 11dB lower)

Table 1: Digital set-top audio decoder nomenclature

Set-top Box Manufacturer	Guide Application	Setup Menu Item	Set-top Mode Selection	Equivalent AC-3 Decoder Operating Mode
Scientfic-Atlanta	*SARA*[11]	*Audio Dynamic Range:*[12]	*Narrow*	*RF Mode*
Scientfic-Atlanta	*SARA*	*Audio Dynamic Range:*	*Normal*	*Line Mode*
Scientfic-Atlanta	*SARA*	*Audio Dynamic Range:*	*Wide*	*Not applicable*[13]
Motorola	*TV Guide*	*Audio \ Audio Output:*	*TV*	*RF Mode*
Motorola	*TV Guide*	*Audio \ Audio Output:*	*Stereo*	*Line Mode*
Motorola	*TV Guide*	*Audio \ Audio Output:*	*Advanced / Heavy*	*RF Mode*
Motorola	*TV Guide*	*Audio \ Audio Output:*	*Advanced / Light*	*Line Mode*
Motorola	*TV Guide*	*Audio \ Audio Output:*	*Advanced / None*	*Line Mode with no DRC*[14]

5. Analog Tier Dialogue Measurement Practices

This section provides guidance in measuring the dialogue level of analog program audio, below 100% modulation of the FM aural carrier. A modulation level of 100% is equivalent to 25 kHz

[10] Set top boxes not listed here require the system to contact the manufacturer for instructions on how to default the set top digital audio decoder to RF mode.
[11] Scientific-Atlanta Resident Application
[12] Some systems may chose to remove the ability for the subscriber to access / change decoder operating modes via the DNCS. However, each set-top should be defaulted to "Narrow" mode.
[13] Use of Wide mode is being deprecated and must not be used.
[14] With the exception of overload protection, dynamic range control metadata (if present within the audio bitstream) is not applied in this mode.

peak deviation as required by the FCC in 47 C.F.R. §73.682. It is important to note, that all measurements must be performed at the output of a calibrated demodulator. In the past, many cable systems have made attempts to perform measurements at the inputs to analog modulation equipment. This practice is not recommended due to the fact that many satellite receivers and IRDs (Integrated Receiver Decoders) have been designed with different maximum output level capabilities which have not been standardized.

Therefore, analog tiered measurements performed according to this recommendation are always relative to 100% modulation (i.e. 25kHz peak deviation). This practice offers a fixed reference point regardless of the capabilities of upstream equipment.

Figure 2: Leq(A) RF Input Dialogue Level Set Correctly at -17dBr

As described earlier in Section 3, it is extremely important for headend personnel to adjust their analog modulation equipment so that the dialogue level for each service is ~ -17dB below 100% modulation. (i.e. 25 kHz Peak Deviation). In Figure 2, the value shown above "meter(C)" indicates the A-weighted "short-term" dialogue level *relative* to 100% modulation. Hence the reason "dBr" is utilized. Note: In North American NTSC systems, 100% modulation is equivalent to 25 kHz peak deviation.

The "peak" value, in Figure 2, is displayed on the right-hand side of the display. This value is the instantaneous un-weighted peak *relative* to 100% modulation. Hence, when this value is at +00 dBr the audio peak deviation is at 100% modulation which is equivalent to 25 kHz peak deviation, and when this value is above +00 dBr the audio peak deviation is beyond 100% as indicated. For example, a peak reading of +06 dBr is equivalent to 200% which is equivalent to 50 kHz peak deviation. Note: The peak measurement will hold the largest (and most recent) peak value for 750 msec. (i.e. The LM100 peak reading display has a 750msec "hold" time) This behavior allows an operator to continuously and easily monitor the largest peaks within the selected program.

While analog audio deviation adjustments are being made, the LM100 must be provisioned in "short-term" measurement mode (indicated by the lower-case "s" in the upper left-hand corner of the dialogue measurement area – see Figure 2). Short-term mode integrates the A-weighted speech level over the previous 10-second period (i.e. it is a moving average). This mode allows the operator to quickly make and monitor audio modulator deviation adjustments.

Figure 3: Leq(A) RF Input Dialogue Level Set Incorrectly to -24dBr

Figure 3 indicates that the current speech level for RF Cable Channel 2 is currently at -24dB below 100% modulation. This value, is 6dB below the -17dBr target value defined in previous sections of this document. Notice that the instantaneous un-weighted peak value is at -6dBr relative to 100% or 25kHz peak deviation. Hence, a simple adjustment to the audio deviation (to bring the speech level up to -17dBr) is all that is needed.

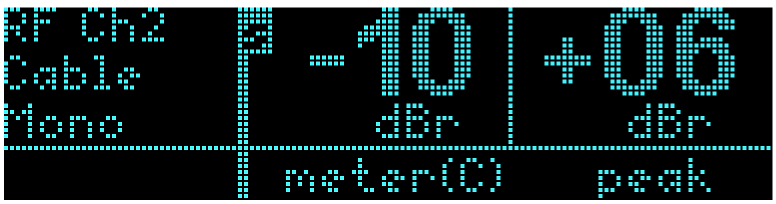

Figure 4: Leq(A) RF Input Dialogue Level Set Incorrectly to -10dBr

Figure 4 indicates that the current speech level for RF Cable Channel 2 is currently at -10dB below 100% modulation. This value, is 7dB above the -17dBr target value defined in previous sections of this document. Notice that the instantaneous un-weighted peak value is at +6dBr relative to 100% or 25kHz peak deviation. Hence, a simple adjustment to the audio deviation (to bring the speech level down to -17dBr) is all that is needed.

6. Digital Tier (including Digital Simulcast) Measurement Practices

The LM100 also includes a digital input that complies with AES3-1992 (SMPTE 276M) specifications (See Figure 11). The connector is a 75-ohm female BNC type and includes a pass-through connector that should be terminated.

When the Input Source selection is set to “digital” (or is currently ‘active’ as defined in Section 8) the LM100 can autodetect the presence of either 2-Channel Linear PCM or a **Dolby Digital (AC-3)** coded bitstream. For cable television applications this input is useful for measuring and monitoring digitally tiered as well as your digital simulcast services. This input requires the use of a digital Set Top Box (STB) to access the demultiplexed Dolby Digital (AC-3) bitstream for presentation to the LM100. Thus, the digital audio output (i.e. S/P DIF) of the digital STB must be connected to the digital input of the LM100 and the STB must be tuned to a digital service. Refer to Figure 8 for a connection diagram.

Note: Some digital set top boxes may require that the operator to choose “Dolby Digital” as their default preference for the digital audio output.

Figure 5: LM100 Display with Digital Input Active

Figure 5 shows the LM100 display with the digital input 'active' and currently measuring a 5.1 channel (3/2L) AC-3 bitstream operating at 448kbps. The 'dialnorm' value, on the right hand side of the display, currently indicates -27dBFS. This value is provisioned by the programmer's audio encoding equipment or within your local digital encoder (for your digital simulcast services) and is meant to indicate the 'long-term' A-weighted average level of spoken dialogue for the digital program.

Hence, the current speech (dialogue) value on the left side (indicated above the word 'meter(C)') (-29dBFS in Figure 5) should agree with the current dialnorm value, over the long-term. You may observe that some channels and programs have different dialnorm values. This practice is OK since there is no standard dialnorm value for 'all' broadcast programming.

The most important point to remember is that the measured long-term dialogue level must match the transmitted dialnorm value for a given program and/or service. That is, the measured value (above the word 'meter(C)' in Figure 5) must be in agreement with the transmitted dialnorm value (above the word 'dialnorm' in Figure 5).

The example displayed in Figure 5 indicates that the measured speech (dialogue) level is 2dB different than the transmitted dialnorm value indicating that this program will be 2dB quieter than a program that has its dialnorm value set to match the dialogue level. In the case where you have control over setting the dialnorm value (e.g. for all of your digital simulcast services) all you would need to do to correct this problem is to change the dialnorm value in your audio encoder to -29. (i.e. you do not need to adjust the input level to the encoder) While digital source (AC-3) measurements are being made, the LM100 must be provisioned in "infinite-term" measurement mode (indicated by the lower-case "i" in the upper left-hand corner of the dialogue measurement area – see Figure 5). Inifinte-term mode integrates the A-weighted speech level over the entire measurement period (i.e. it is a long-term integrated average). This mode allows the operator to generate accurate long-term dialogue measurements to be used for setting the dialnorm value in your audio encoders.

In order to set the dialnorm value for each of your digital simulcast services properly[15], please consider the following measurement practices:

[15] For digital simulcast services, the dialnorm value is set in your local encoder. In some cases, this parameter is accessed in an "advanced" settings menu.

1. All measurements must be made in the digital domain. (PCM or pre-compressed AC-3) This ensures you are utilizing a ‘fixed’ loudness measurement reference. Do not attempt to measure (and derive the dialnorm value) from the baseband analog inputs.
2. The LM100 must be provisioned in “infinite-term” measurement mode (indicated by the lower-case “i” in the upper left-hand corner of the dialogue measurement area – see Figure 6). Inifinte-term mode integrates the A-weighted speech level over the entire measurement period (i.e. it is a long-term integrated average).
3. The LM100 must have “Dialogue Intelligence” enabled.
4. The LM100 “measurement channel” must be set to “Center”.
5. “Reset” the measurement at the beginning of the measurement period. (i.e. for each channel being analyzed)
6. For digital simulcast applications, it is recommended to take a measurement over an extended period of time[16]. This practice will allow you to come up with an accurate long-term speech level for the channel you are analyzing. At the end of the measurement period, the average level of speech (integrated over the entire measurement period) will be displayed above the word ‘meter(C)’ in the LM100 display. This value can then be utilized as the dialnorm value in your (audio) simulcast encoder. Hence, if the long-term measured speech level is -22dBFS then -22 shall be used as the dialnorm value in your simulcast (audio) encoder.

Figure 6: Digital Simulcast Measurement (showing a nearly correct dialnorm value)

7. DPI & VOD Measurement and Encoding Practices

Proper encoding of advertising spots must follow the same practices used for digitally tiered measurements (and encoding) explained earlier. Many people ask the following question when it comes to encoding spots and/or programs: “Why isn’t there a standard?” When actually there **is** a standard[17] for these types of services/programs too. However, this fact is often overlooked. The standardized method involves setting the dialnorm value properly for each spot/program you encode. Only then will the spot(s)/program(s) created play back at the correct level in both Line and RF decoder operating modes.

[16] The actual measurement period can vary from a few hours (for a channel/service that maintains consistent loudness levels from day to day) to multiple hours (for channel/services that do not maintain a consistent loudness level from day to day or intra-day). This is easily determined by logging the speech loudness values (with the LM100 Window Remote Application) during your measurements and analyzing the ‘logged’ data for consistency.

[17] For DTV, the FCC mandates the use of ATSC A/53D which in Section 5.5 states: “The value of the dialnorm parameter in the AC-3 elementary bit stream shall indicate the level of average spoken dialogue within the encoded audio program.” The word “shall” (above) denotes a mandatory provision of the ATSC A/53D standard.

The largest problem today is that many of the facilties that encode spots/programs locally or nationally do not set the dialnorm value (in the audio encoder) to agree with the ACTUAL measured level of dialogue contained in the spot or program. In other words, they leave the dialnorm value at the default setting which in many cases is -27 or -31.

In order to set the dialnorm value properly, please consider the following measurement practices:

7. All measurements must be made in the digital domain. (PCM or pre-compressed AC-3)
8. The LM100 must be provisioned in "infinite-term" measurement mode (indicated by the lower-case "i" in the upper left-hand corner of the dialogue measurement area – see Figure 7). Inifinte-term mode integrates the A-weighted speech level over the entire measurement period (i.e. it is a long-term integrated average).
9. The LM100 must have "Dialogue Intelligence" enabled.
10. The LM100 "measurement channel" must be set to "Center".
11. "Reset" the measurement at the beginning of the measurement period.
12. For advertising spots, take a measurement over the entire spot. At the end of the spot the average level of speech (integrated over the entire measurement period) will be displayed above the word 'meter(C)' in the LM100 display. This value can then be utilized as the dialnorm value in your audio encoder. Hence, if the measured speech level is -22dBFS then -22 shall be used as the dialnorm value in your audio encoder before starting your encoding pass. For VOD and other long-form programming, taking measurements over several "representative sections"[18] of the program often generate good results in-lieu of measuring over the entire program (from beginning to end).

Figure 7: Digital Program Measurement (showing incorrect dialnorm value)

[18] The term "representative sections" indicates shorter sections of the program (being measured) that contain spoken dialogue levels that are 'representative' of the speech levels found throughout the entire program.

8. LM100 Connection Recommendations

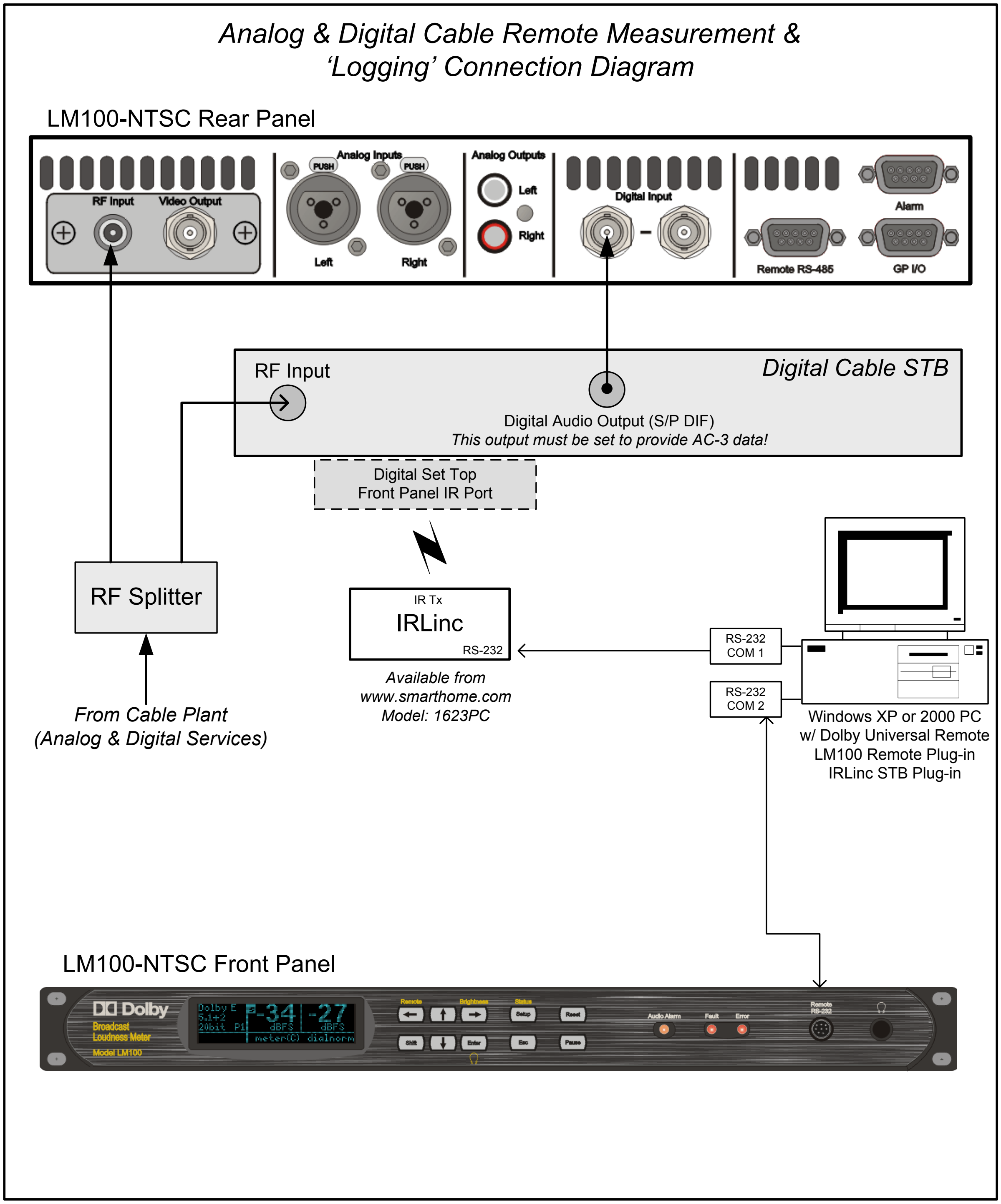

Figure 8: LM100 Connection Diagram

Figure 8 shows the recommended connections between the cable plant, digital set top box and the LM100. The LM100 includes the ability to autodetect (via user selection) the presence of a Dolby Digital (AC-3) bitstream (on the digital input) when the RF input is currently active. This special mode of operation is selected from within the 'setup' menu. Once the setup menu is active (by pressing the front panel 'Setup' button), select the 'Input Control' menu then select the 'Input Source' sub-menu and choose 'RF/Digital'.

Operating in this mode (and with the LM100 connected as is Figure 8) allows the operator to easily measure and monitor both analog and digital services in their respective domains that offer fixed measurement references. In this mode, analog services are selected via the LM100s front panel (see Section 9) while digital services are selected via the digital set top box. Hence, when the digital set top box is tuned to a digital service, the digital audio output on the set top box will output a Dolby Digital (AC-3) bitstream where the LM100 will automatically detect and analyze the digital audio (AC-3) bitstream. To return to analyzing the analog services, the operator must tune the set top to any non-digital service.

Note: It is **not recommended** that the baseband (line) or RF remodulator outputs of digital set top boxes be utilized for measurement and/or analysis. Since these outputs often have a volume control associated with them.

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9. Appendix A - Dolby Laboratories LM100 Broadcast Loudness Meter

The LM100 is the only loudness meter specifically developed to measure human speech contained within broadcast programming. The unique algorithm named 'Dialogue Intelligence' allows the LM100 to automatically base its measurement on the portions of the input signal that primarily contain the characteristics of dialogue. Within broadcast programming the dialogue (or speech) levels of a program are especially important when considering how to determine the subjective loudness of a program. Research shows that most listeners at home use their volume controls in an effort to keep the dialogue levels uniform as they switch through multiple channels and as they listen through program-to-program and/or program-to-commercial transitions. It therefore follows that if listeners at home base their loudness judgment on the level of dialogue, then a loudness meter would certainly benefit from an algorithm that mirrors this behavior. Dialogue Intelligence™ in the LM100 provides exactly that feature.

This powerful feature gives users the ability to easily quantify the level of dialogue within broadcast programs no matter what their skill level may be. By combining Dialogue Intelligence with extensive logging and alarm capabilities, the LM100 opens up the possibility of automated measurement, QC, and control.

When you first enable Dialogue Intelligence, the LM100 takes a few moments to analyze the input signal before confirming that dialogue is or is not present. During this analysis, a "listening" icon displays, as shown in Figure 9. When the analysis is complete, it displays either '`No Dialogue`' or a dialogue-based measurement value.

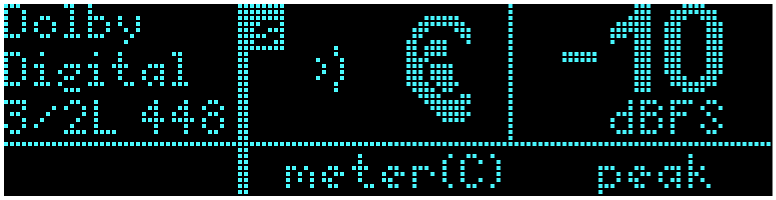

Figure 9: 'Listening Icon' Display before Dialogue Intelligence Computes

Remember, when Dialogue Intelligence feature is enabled, the measurement is based solely on the portions of the input signal recognized as having the characteristics of speech and portions of the input signal that do not primarily contain the characteristics of dialogue are not included in the measurement value.

Note: The Dialogue Intelligence algorithm is designed to return a dialogue-based measurement value only when this input signal primarily contains the characteristics of dialogue. Therefore, on rare occasions, the algorithm may ignore a section of the program that contains dialogue but may also be coincident with other types of signals and spectra, such as music or effects. This behavior increases the accuracy and confidence of the dialogue-based measurement value.

The LM100-NTSC also includes an 800 MHz analog RF tuner module that can be controlled via the front panel (i.e. while the LM100 is in the main status menu, **Shift**, 🡅 (up-arrow) increases the selected RF channel number and **Shift** 🡇 (down-arrow) decreases the selected RF channel number), rear panel GP I/O connector (contact closure) or serially via the RS-232 (front panel) or RS-485 (rear panel) ports. **Hence, any "in the clear" analog channel between channel 2 and 125 can be monitored and/or measured without the need for a separate device for each channel in your plant.**

The RF Tuner module **supports both monophonic and BTSC encoded (i.e. stereo) services**. The audio service type (i.e. Stereo or Mono) is indicated on the left-hand side of the main status screen as shown in Figure 10.

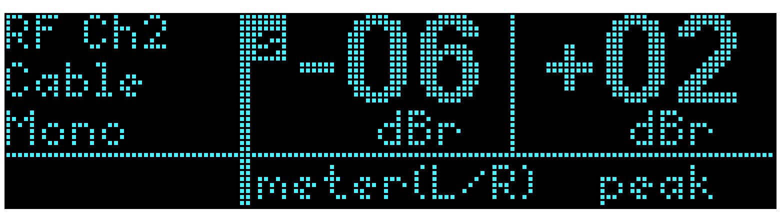

Figure 10: LM100 Display while RF Input is Active (mono channel shown)

Two RF channelization modes are supported; Off-Air and Cable. When the LM100s RF Tuning Mode is set to "Cable" the default channelization utilized is compliant with the EIA-542 "standard" frequency plan however; HRC and IRC frequency plans can be also be selected, if applicable. The user can also select and measure SAP (Secondary Audio Programming) if present within the BTSC encoded signal.

The tuner module input is a standard F-Type (female) connector with an internal 75-ohm termination. Also note the tuner module includes a composite video output available on a female BNC connector. This output allows confidence monitoring of the video for the currently tuned program. Refer to the left-hand side of Figure 11.

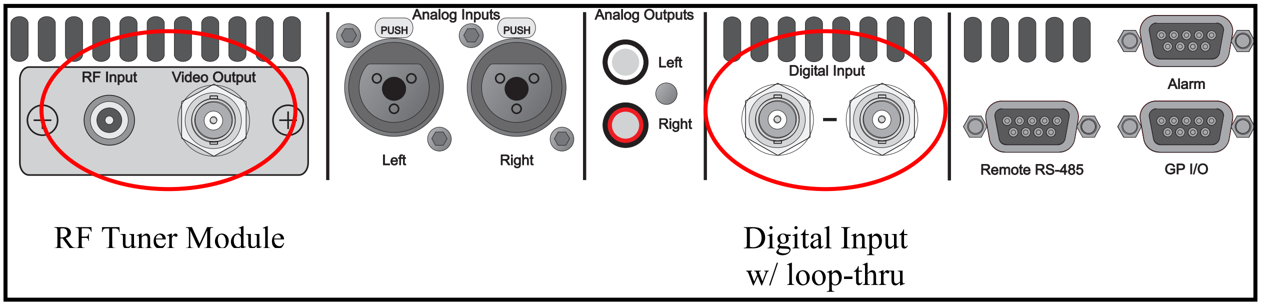

Figure 11: LM100 Rear Panel

The LM100 also includes a digital input that complies with AES3-1992 (SMPTE 276M) specifications (See Figure 11). The connector is a 75-ohm female BNC type and includes a pass-through connector that should be terminated.

When the Input Source selection is set to "digital" (or is currently 'active' as defined in Section 8) the LM100 can autodetect the presence of either 2-Channel Linear PCM or a **Dolby Digital (AC-3)** coded bitstream. For cable television applications this input is useful for measuring and monitoring digitally tiered as well as your digital simulcast services. This input requires the use of a digital Set Top Box (STB) to access the demultiplexed Dolby Digital (AC-3) bitstream for presentation to the LM100. Thus, the digital audio output (i.e. S/P DIF) of the digital STB must be connected to the digital input of the LM100 and the STB must be tuned to a digital service. Refer to Figure 8 for a connection diagram.

10. Appendix B – Listeners Comfort Zone[19]

There is presumably some loudness range, call it the Comfort Zone, within which a listener will accept loudness changes within and between programs. Assuming further that the non-speech elements of the programs have been appropriately "balanced" around the speech elements, listeners will not be annoyed by the natural changes in loudness that occur during programs if the speech elements fall within their individual Comfort Zone.

To the best of our knowledge, the magnitude of this Comfort Zone has never been determined. A series of subjective experiments was undertaken to determine the range of loudness levels that define the Comfort Zone.

The basic method was to present listeners with a Reference program segment whose loudness they adjusted to their own comfortable listening level. Thy were then asked to adjust the

[19] Excerpt from: Riedmiller, Lyman & Robinson – 115th AES Convention Paper 5900 – "Intelligent Program Loudness Measurement and Control: What Satisfies Listeners?

loudness of a test segment until it was louder than the reference item, but still acceptable, an then to adjust the same item until it was softer than the reference item, but still acceptable. These thresholds were taken as defining the listener's Comfort Zone. Listeners were also asked to adjust the loudness of the test item to several other thresholds, both to investigate these points and to allow for the necessary randomization within in the experiment. A useful objective loudness measurement should be able to put most programs into most listeners' Comfort Zone most of the time. Obviously, it is impossible to do this all the time for all listeners; both the impression of loudness and the comfort zone are individual opinions. It is not always desirable to make all programs equally loud either. A rock concert or some parts of an action film would seem silly if they were not louder than a current affairs discussion. The point of this experiment was to develop criteria for the requirements of a useful loudness meter, and to give some guidance to broadcasters about what the Comfort Zone of their listeners might be as well as to let them know what their listeners might find objectionable.

The experimental scenario was to place listeners in a typical listening or viewing situation during which the program would switch from one type of program to another. Subjects were presented with several paired, monophonic Reference and Test program samples, reproduced by a single loudspeaker in front of them, in a typical listening room.

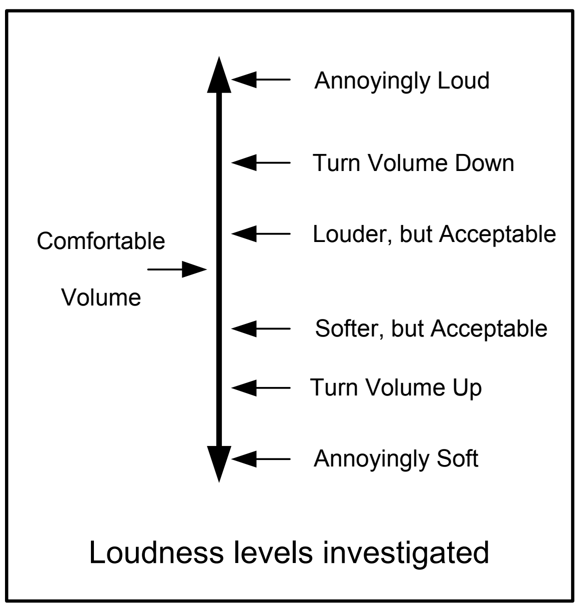

Figure 12: Loudness Levels Investigated

They were instructed to adjust the master playback level until the Reference item was reproduced at what they considered to be a "comfortable volume". The experimenter then asked them to set the Test volume control ("volume" is familiar to most listeners) to one of the six points shown in Figure 12. The questions were asked in random order, and the order of presentation of the pairs of reference and test items was randomized between subjects.

The pairs of program samples were taken from the library of equal subjective loudness samples developed from the loudness matching experiment, so the gain offset the subjects applied to the test item in response to the questions from the experimenter were a direct measurement of the subjects' comfort zone and other critical loudness levels.

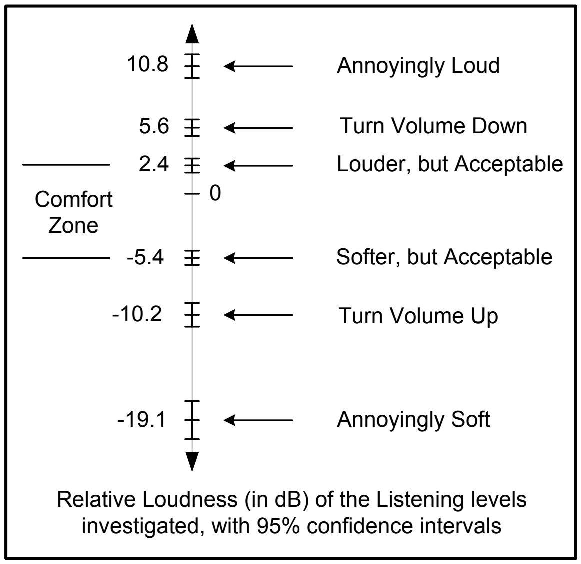

Figure 13 : Comfort Zone (Listener Level Tolerances)

Figure 13 shows the results of the Comfort Zone tests. The interesting aspect is that the results make it quite clear why television broadcasters (and others) have been plagued by complaints of "loud ads" for so long. An increase of two to three dB in subjective loudness is enough to move a program out of the typical listeners Comfort Zone, and toward the point at which they would like to turn the volume down. There is much more latitude available on the softer side of the "comfortable volume" point (shown here as "0"). One point should be mentioned, however. The ambient noise level in the listening room used for the tests was quite low; similar to a suburban living room on a tranquil evening. Since the "Annoyingly Soft" point can reasonably be expected to fall somewhere above the ambient noise level in the listening environment, the figure of –19.1 dB may depend on the ambient noise level. The other points are far enough above the ambient that they should not be affected.

11. Appendix C – Suggested Measurement Points

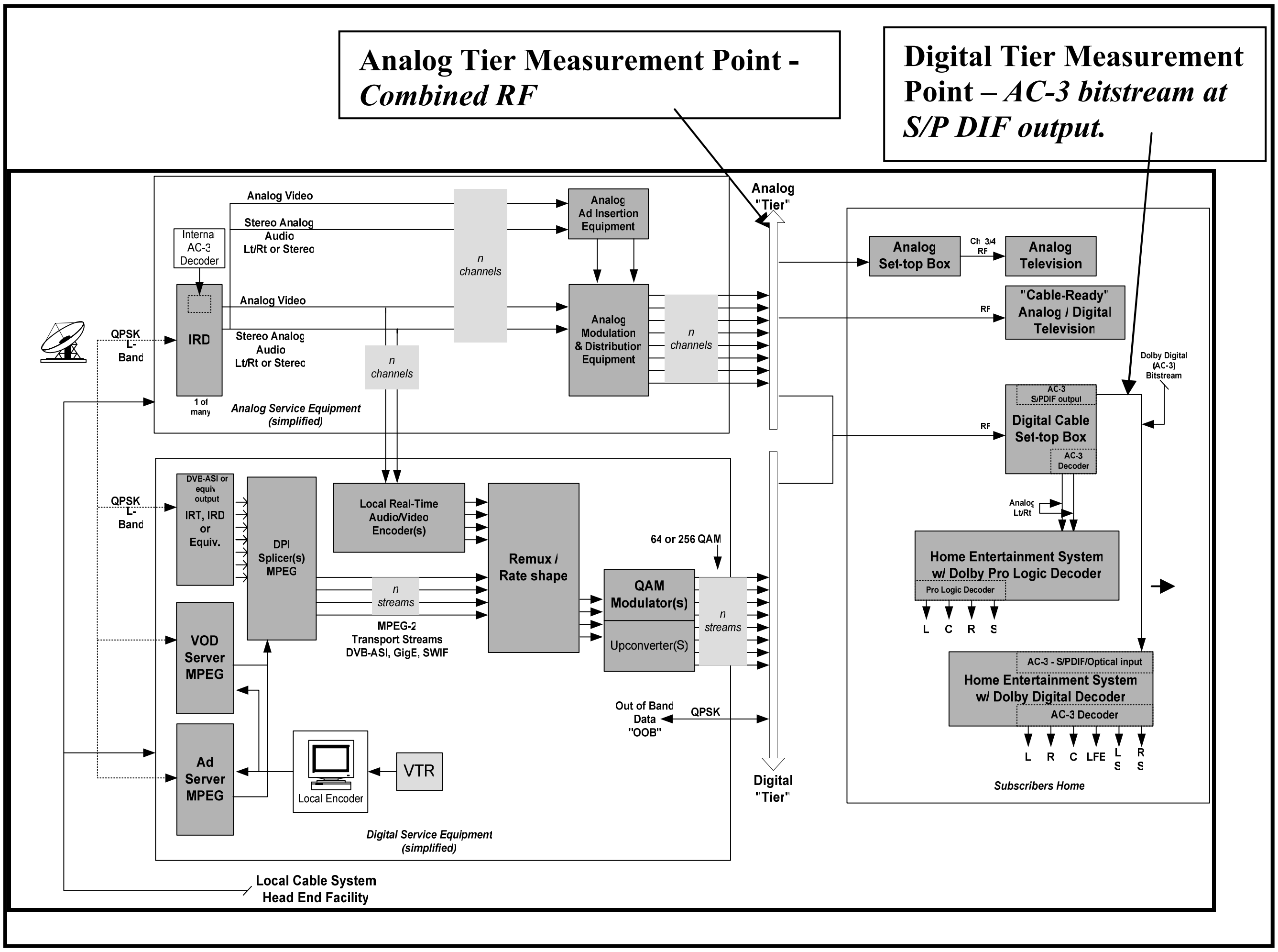

Figure 14: Suggested Measurement Points