Model 9670

LevelTrack Audio Loudness Control
Automatic Gain Control Software
User Guide for Models:

7555, 7660, 7660-XV, 9550, 9550-XA, 9600 and 9600-XV

Family of 3G/HD/SD Embedders, Disembedders, Data Inserters and Video Processing Frame Synchronizers

Revision 1.5

ENSEMBLE DESIGNS

Purveyors of Fine Video Gear—Loved by Engineers Worldwide

Clearly, Ensemble wants to be in the broadcast equipment business. It's so rare anymore to find a company of this caliber that has not been gobbled up by a large corporation. They are privately held so they don't have to please the money people. They really put their efforts into building products and working with customers.

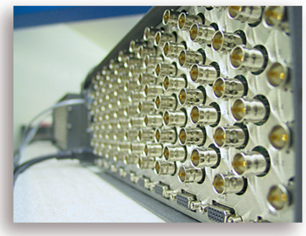
I'm really happy with the Avenue products and Ensemble's service, and even more important my engineers are happy. We've continued to upgrade the product and add more cards. We will be rebuilding our production control room and we will use Avenue again.

~ *Don McKay, Vice President Engineering, Oregon Public Broadcasting*

Who is Ensemble Designs?

By Engineers, For Engineers

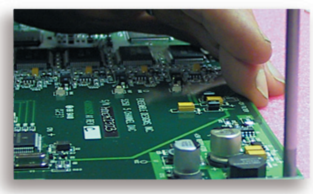
In 1989, a former television station engineer who loved designing and building video equipment, decided to start a new company. He relished the idea of taking an existing group of equipment and adding a few special pieces in order to create an even more elegant ensemble. So, he designed and built his first product and the company was born.



Avenue frames handle 270 Mb/s, 1.5 Gb/s and 3 Gb/s signals, audio and MPEG signals. Used worldwide in broadcast, mobile, production, and post.

Focused On What You Need

As the company has grown, more former TV station engineers have joined Ensemble Designs and this wealth of practical experience fuels the company's innovation. Everyone at the company is focused on providing the very equipment you need to complete your ensemble of video and audio gear. We offer those special pieces that tie everything together so that when combined, the whole ensemble is exactly what you need.



We're focused on processing gear– 3G/HD/SD/ASI video, audio and optical modules.

Notably Great Service for You

We listen to you – just tell us what you need and we'll do our best to build it. We are completely focused on you and the equipment you need. Being privately held means we don't have to worry about a big board of directors or anything else that might take attention away from real business. And, you can be sure that when you call a real person will answer the phone. We love this business and we're here to stay.

Come on by and visit us. Drop in for lunch and a tour!

Bricks and Mortar of Your Facility

The bricks and mortar of a facility include pieces like up/downconverters, audio embedders, video converters, routers, protection switches and SPGs for SD, HD and 3Gb/s. That's what we're focused on, that's all we do – we make proven and reliable signal processing and infrastructure gear for broadcasters worldwide, for you.

Shipped with care to television broadcasters and video facilities all over the world.

Contents

LevelTrack™ Loudness Control: An Audio Loudness AGC Software Option for Avenue Systems

Overview

The LevelTrack Audio Loudness Control Automatic Gain Control (AGC) Software option adds an operator configurable audio level management system to Avenue signal processing modules. LevelTrack Loudness Control will correct mismatched audio levels between different program sources or segments within a program. Errors of this type are regrettably common due to inconsistencies between different providers and program elements.

LevelTrack Loudness Control will automatically monitor the levels in up to 16 audio channels. Based upon the history in each channel, LevelTrack Loudness Control applies gradual changes to prevent the audio level from dropping below or exceeding user programmable thresholds. The operator can apply this automatic level control to an individual channel, stereo pair, or a Dolby™ Surround group. By adjusting the overall level of the signal rather than masking the errors with compression, LevelTrack Loudness Control will not upset the internal dynamics of the program material.

LevelTrack Loudness Control provides operator control over the following parameters:

- audio target level and spread
- transition time
- maximum gain and attenuation

This flexibility allows the operator to customize LevelTrack Loudness Control to suit the specific audio level challenges in a particular installation. The operator can adjust all of these parameters through the Avenue Control System. LevelTrack Loudness Control operates downstream of the manual audio level adjustments that are already provided in Avenue modules. This allows the automatic feature to assist the operator when needed without needing to be enabled or disabled.

The LevelTrack Loudness Control product is available through a software license and does not require any additional hardware.

The following Avenue Modules can be used with LevelTrack Audio Loudness Control AGC Software:

- 7555 HD/SD Video Processing Frame Synchronizer
- 7660 HD/SD Embedder, Disembedder and Data Inserter
- 7660-XV HD/SD Embedder, Disembedder and Data Inserter – Additional Video Outputs
- 9550 3G/HD/SD Video Processing Frame Synchronizer
- 9550-XA 3G/HD/SD Video Processing Frame Synchronizer – Additional Audio Outputs
- 9600 3G/HD/SD Embedder, Disembedder and Data Inserter
- 9600-XV 3G/HD/SD Embedder, Disembedder and Data Inserter – Additional Video Outputs

Note: Modules require software version 2.2.10 or higher and Avenue PC must be version 2.0.15 or higher. Software upgrades are free and can be downloaded from our website. Please see details in the “Software Requirements” on page 6.

Software Requirements

Make sure you have installed software version 2.2.10 or higher in your Avenue module (7555, 7660, 7660-XV, 9550, 9550-XA, 9600, 9600-XV) and the latest version of Avenue PC (version 2.0.15 or higher) in order for LevelTrack Audio Loudness Control AGC Software to function properly. Software updates are free. You must have a valid serial number for your module and for Avenue PC in order to download the software.

To download the latest software, go to the following URLs:

http://www.ensembledesigns.com/support/avenue-support/avenue-software

http://www.ensembledesigns.com/support/avenue-support/avenue-pc-software

Configuring LevelTrack Audio Loudness Control AGC

Configuring LevelTrack Audio Loudness Control AGC Software involves the menus in the table immediately below. Each menu is discussed in greater detail in the subsequent pages in this user guide.

While the menu examples in this document are taken from the Avenue 9600 Module, these menus and controls are applicable to other Avenue Modules that are compatible with LevelTrack Audio Loudness Control AGC Software.

License Menu	Use this menu to enter the key provided by Ensemble Designs in order to activate LevelTrack Audio Loudness Control AGC Software. See page 7 for details.
Aud Cfg Menu	Use this menu to establish settings for the controls **Meter Mode** (LKFS or dBFS), **Meter Position** (pre or post fader) and **LKFS or dBFS Average Time**. See page 8 for details.
Aud AGC Menu	Use this menu to configure most of the settings for LevelTrack Loudness Control, including the **Audio Target Level**, **Spread** and **Transition Time**. See page 10 for details.
Mix 1:4, Mix 5:8, Mix 9:12, Mix 13:16 Menus	Use these menus to configure how you want the mixer channels to work with each other and to **Enable AGC** for specific channels or associated channels, such as for stereo or Surround Sound. See page 13 for details.

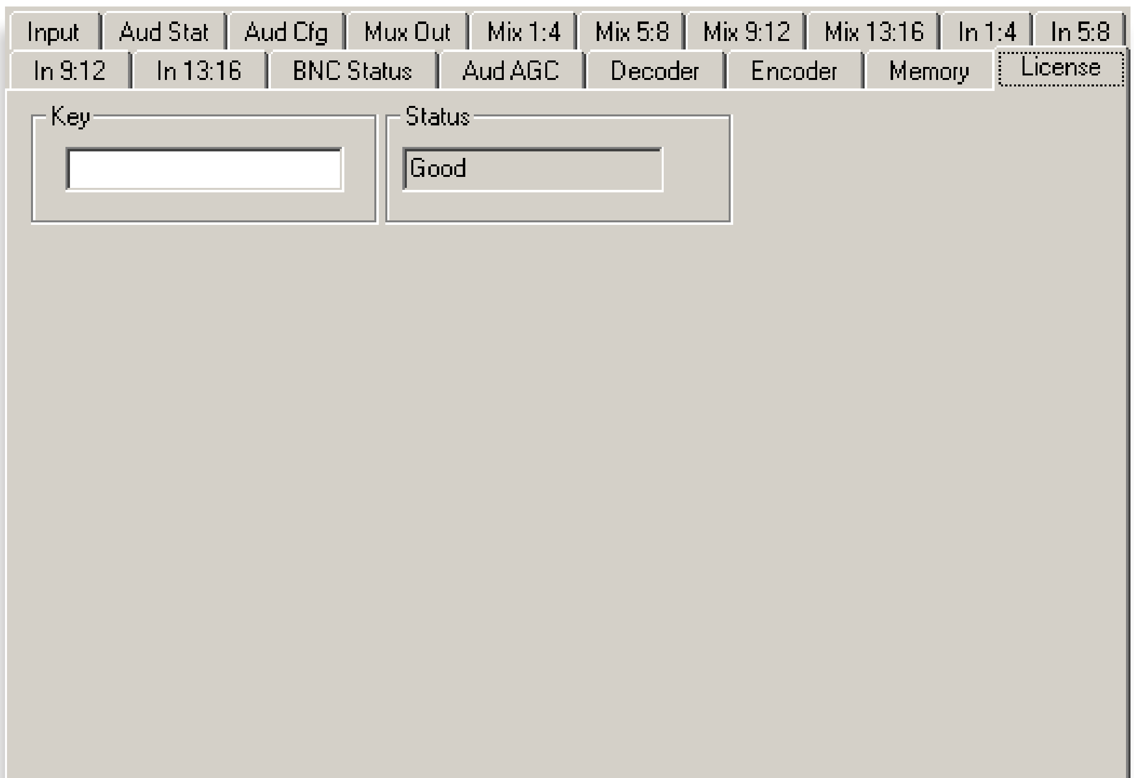
Entering the Key

1. Launch Avenue PC software. (Alternately, the Avenue Touch Screen can be used.)

2. From Avenue PC, select the frame and then the module to display the module's menus.

3. Select the module's **License** menu. The **License** menu displays.

4. Enter the key provided by Ensemble Designs into the **Key** field, then press **Enter** on your keyboard. If the key you entered is valid, the **Status** field will display "Good." If it is invalid, the **Status** field will display "Invalid." If you do not enter a key and press **Enter**, the **Status** field will display "None."



***License** Avenue PC Menu*

Aud Cfg Menu

Within the Audio Configuration menu, the **Meter Mode**, **Meter Position**, and **Average Time** controls, discussed on this page, relate specifically to the LevelTrack Loudness Control functionality. For detailed information regarding the other controls in this menu, please refer to your specific module manual (7555, 7660 and 7660-XV, 9550 and 9550-XA, 9600 and 9600-XV).

Meter Mode

Select between LKFS and dBFS. This selection determines the method by which the audio is analyzed and measured, and will impact how Audio AGC behaves.

- **LKFS** – LKFS (Loudness K-weighted relative to Full Scale) is a loudness amplitude level based on the ITU-R BS.1770 Loudness Measurement Method. It is a scale for audio measurement similar to VU or Peak, but rather than measuring gain, it measures perceived loudness. Based on a complex algorithm, this method takes into account audio processing that increases perceived loudness without increasing gain. LKFS is the measurement method required to comply with the Calm Act.

- **dBFS** – dBFS (Decibels relative to Full Scale) is a more traditional method used for measuring audio volume. For more information on decibels and dBFS, please refer to the "Glossary" on page 23.

Meter Position

The Meter Position effects both the **Final Average (yellow line)** on the AGC chart and the audio input level for the **9690 Audio Compliance and Monitoring Software**. The Meter Position is factory set to Post.

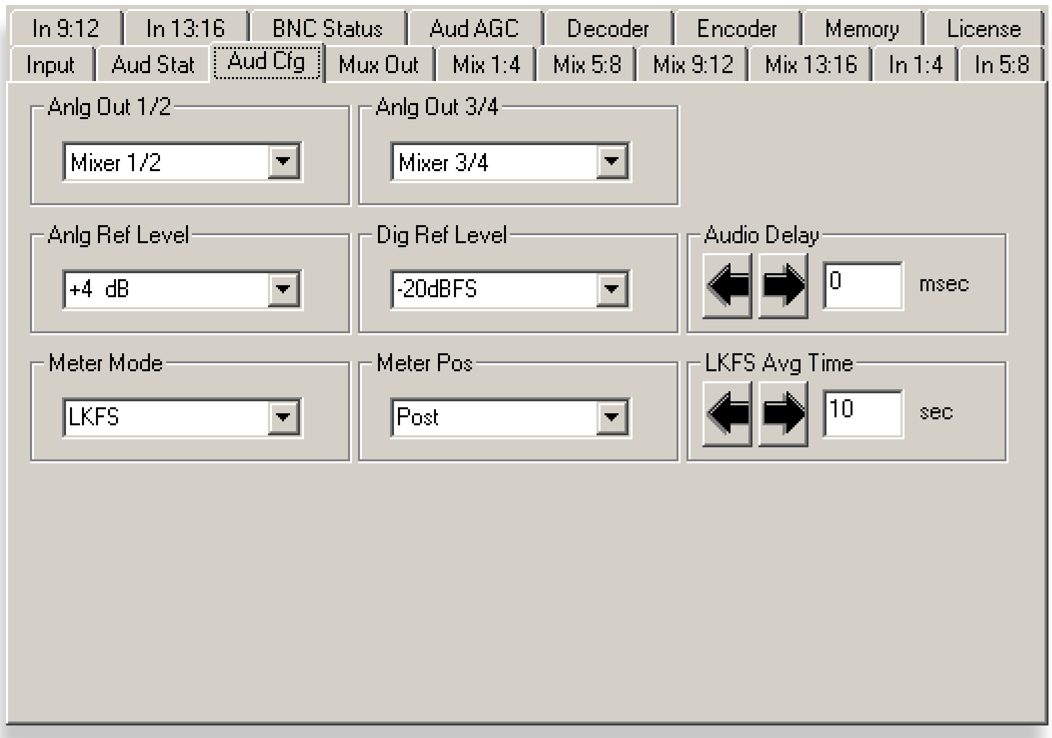
- **Pre (pre-fader)** – When Pre is selected, the **Final Average (yellow line)** on the AGC chart will not reflect any manual adjustments made in the mixer to the gain level of the channel being monitored. Similarly, the chart and recording in the **9690 Audio Compliance and Monitoring Software** will reflect the audio input level coming from your source prior to any gain or attenuation being applied in the mixer.

- **Post (post-fader)** – When Post is selected, the **Final Average (yellow line)** on the AGC chart reflects manual adjustments made in the mixer to the gain level of the channel being monitored. Similarly, the chart and recording in the **9690 Audio Compliance and Monitoring Software** reflect the audio input level coming from your source after any gain or attenuation is applied in the mixer.

Average Time

Use this control to set the amount of time used to determine the LKFS or dBFS average. 10 seconds is a typical setting.

> **Note**: Although the control is labeled "LKFS Avg Time," if you have selected dBFS as your meter mode, the LKFS Avg time control will actually be reflecting dBFS Avg time.



***Aud Cfg** Avenue PC Menu*

Aud AGC Menu

The following is a detailed description of the LevelTrack Audio Loudness Control AGC controls and how they are used:

AGC Master

- **Off** – When set to Off the LevelTrack Loudness Control AGC functions are turned off. At the moment that LevelTrack Loudness Control is switched off it will smoothly reduce the gain or attenuation (if any) that it had been applying.

- **On** – When set to On, the LevelTrack Loudness Control system engages. It will use the measured dBFS or LKFS of the incoming signal to determine how much gain or attenuation should be applied.

Final Gain

This status indicator shows how much correction, either gain or attenuation, the LevelTrack Loudness Control system is applying.

Silence Limit

0 to -70, factory set to -40 LKFS.

Use this control to establish the value for what is considered to be silence. For example, when set to the value of -40 LKFS, levels that are at and below that value are treated as silence.

Target Level

0 to -50, factory set to -24 LKFS.

The Target Level setting establishes the target output audio level. The LevelTrack Loudness Control AGC function will automatically apply gain or attenuation to the signal to bring it within the range defined by the Target Level and the Spread.

Note: LKFS is interpreted as the inverse of Dialnorm. For example, if your goal is to output Dialnorm 24, set your Target Level at -24. For more detailed information regarding Dialnorm and the Calm Act, see “Ensemble White Paper on Dialnorm - Broadcaster Compliance with the Calm Act” on page 20.

Spread

0 to 50, factory set to 1 LKFS.

Set the Spread from x to x. The Spread indicates how far above and below the Target Level you want to allow the AGC to go. A typical setting is 1. If, for example, the Target Level is set at -24 LKFS, and the Spread is set at 1, the AGC will aim to keep the output signal between -25 and -23 LKFS.

Transition Time

0.5 sec to 30 sec, factory set to 3 seconds.

This setting controls how rapidly LevelTrack Loudness Control will make adjustments once it determines that a change is needed.

Max Atten

0 dB to -12 dB, factory set to -12 bB.

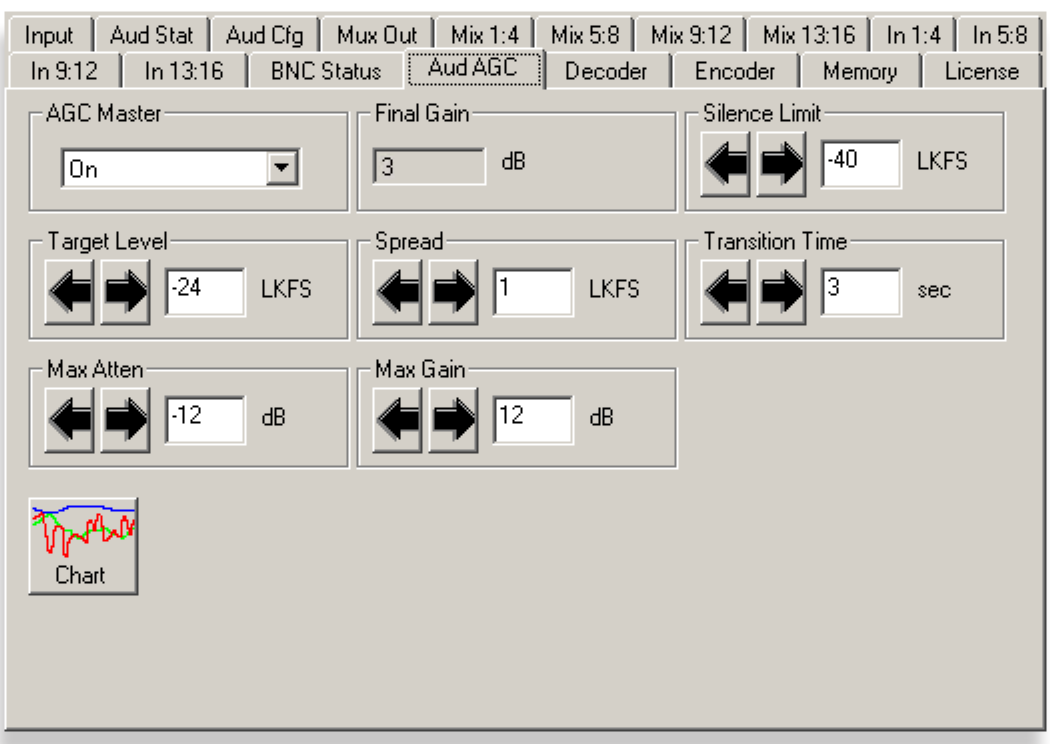
This control sets the maximum amount of attenuation that LevelTrack Loudness Control can use to reduce audio levels.

Max Gain

0 dB to +12 dB, factory set to 12 bB.

This control sets the maximum amount of gain that LevelTrack Loudness Control can apply to the input in order to raise audio levels.

Taken as a whole, these controls provide tremendous flexibility in both how LevelTrack Loudness Control AGC is configured and in how audio is perceived by the listener.



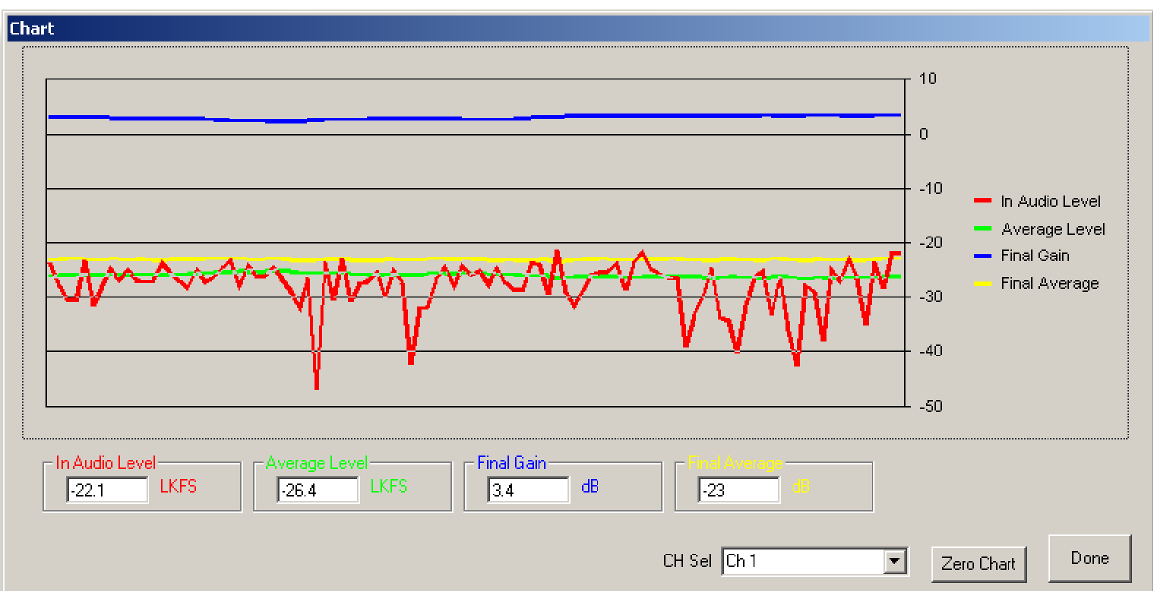
***Aud AGC** Avenue PC Menu*

Chart

Click the **Chart button** to view a visual representation of AGC behavior on a channel-by-channel basis. The chart represents the most recent two-minute span of time for analysis performed on the channel selected in the **CH Sel** drop-down menu.

- **In Audio Level (red line):** The red line represents the level of the audio signal as it enters the Avenue module, prior to being processed by AGC.
- **Average Level (green line):** The green line represents an averaging of the incoming audio signal level.
- **Final Gain (blue line):** The blue line represents the Final Gain expressed in terms of decibels (dB). This shows how much the AGC is adjusting the level of the audio signal based on the configuration parameters specified in the **Aud AGC** menu.
- **Final Average (yellow line):** The yellow line represents the final corrected output, calculated from the Average Level and the Final Gain. The yellow line reflects manual adjustments made to the gain level on the mixer for the channel being charted, provided that the **Meter Position** is set to **Post** on the **Aud Cfg** menu.
- **CH Sel** drop-down menu: LevelTrack Loudness Control automatically monitors the levels in up to 16 audio channels. From the drop-down menu, select the channel for which you want to view the LevelTrack Loudness Control AGC behavior.

Note: The Chart's graph lines remain active as long as you are looking at the corresponding module on Avenue PC. However, if you keep the chart window open, and then select a different Avenue module through Avenue PC, the chart's graph lines will go flat. AGC is still active, however, until it is turned off in the **AGC Master** drop-down menu.



Aud AGC *Chart*

Note: The numerical indicators below the chart are labeled as LKFS. If you have selected dBFS as your Meter Mode in the **Aud Config** menu, the values will be in dBFS despite the labels.

Mix 1:4, Mix 5:8, Mix 9:12, Mix 13:16 Menus

Combinations of Input Channels

One common method of working with the mixer is to put the signals through unchanged, using the mixer only to indicate out bus assignments. However, you can also associate channels with one another by making a selection from the **Mix Mode** drop-down control, discussed on the next page.

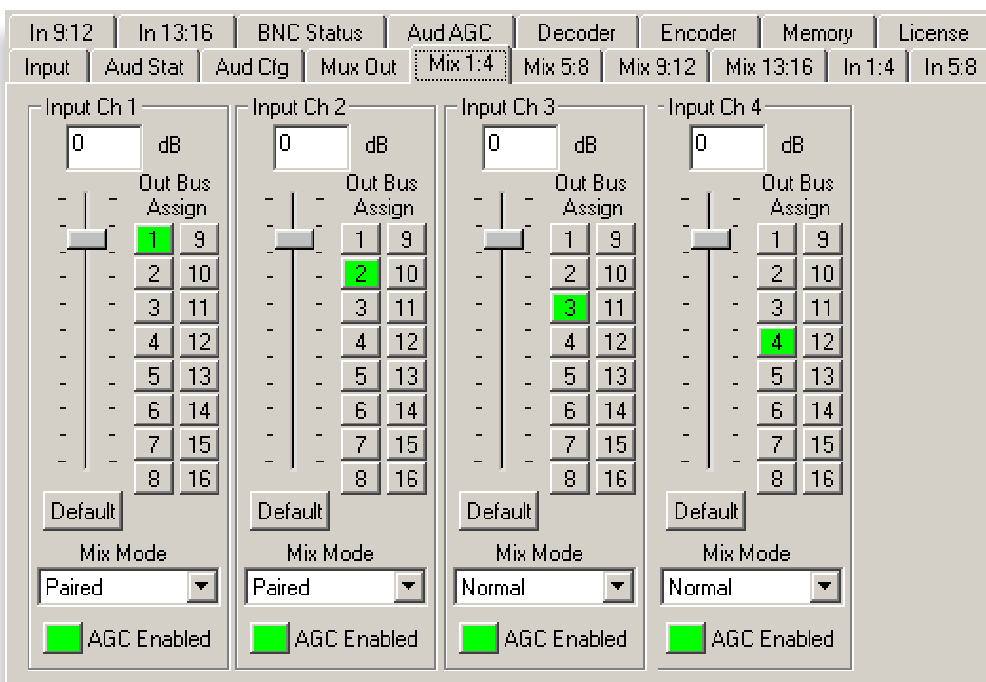
Any particular channel can be independent or it can be tied to other channels. Channels may be paired or stereo, or grouped for Dolby Surround Sound 5.1 or 7.1. When multiple channels are associated together, LevelTrack Loudness Control AGC processing (if enabled) takes into account any channel pairs or Surround Sound groupings.

For modules that have a Dolby E decoder model, any 8 channels may take the input from the Dolby E decoder, leaving 8 remaining input channels to assign.

AGC Enabled

Enable AGC for any mixer channel by selecting the **AGC Enabled** box located at the bottom of each channel. Each **AGC Enabled** box displays green when enabled and grey when disabled. All channels that are AGC enabled will be impacted by the AGC.

Note, however, that the **AGC Enabled** control will have no effect unless AGC is first engaged. To turn on the AGC function, select **On** from the AGC Master control in the **Aud AGC** menu as discussed on page 10.



Mix 1:4 *Avenue PC Menu*

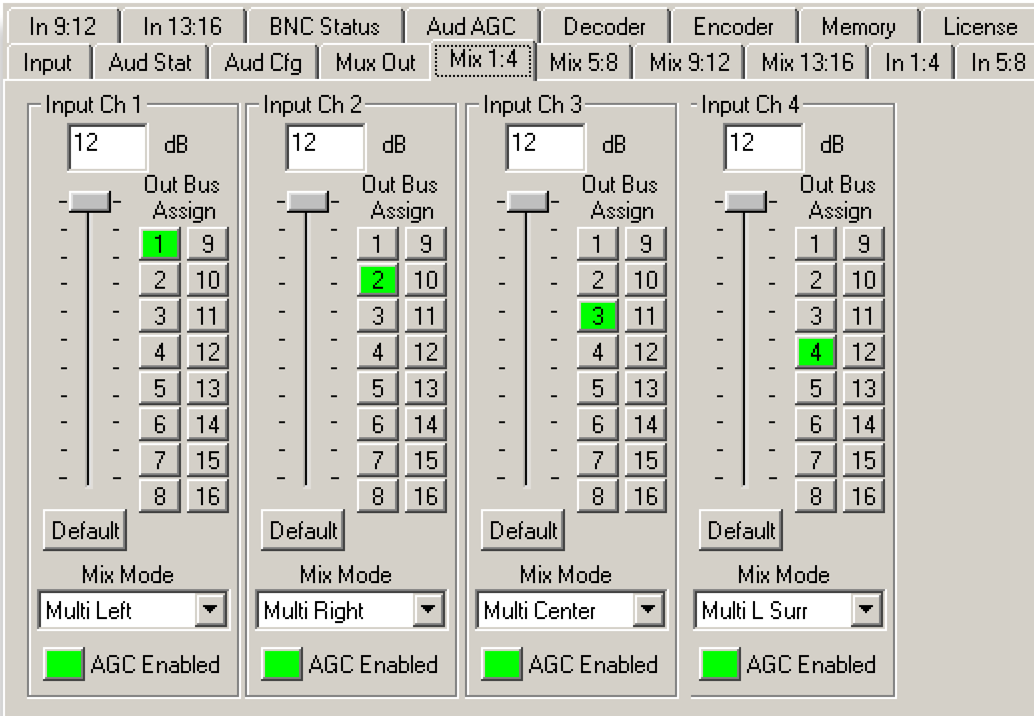
Note: At this time, the mixer menus (Mix 1:4, Mix 5:8, Mix 9:12, Mix 13:16) do not function with the Avenue Touch Screen interface. A pending software update will enable this control. All mixer functionality is currently available through the Avenue PC interface. Please be sure you have Avenue PC version 2.0.15 or higher installed.

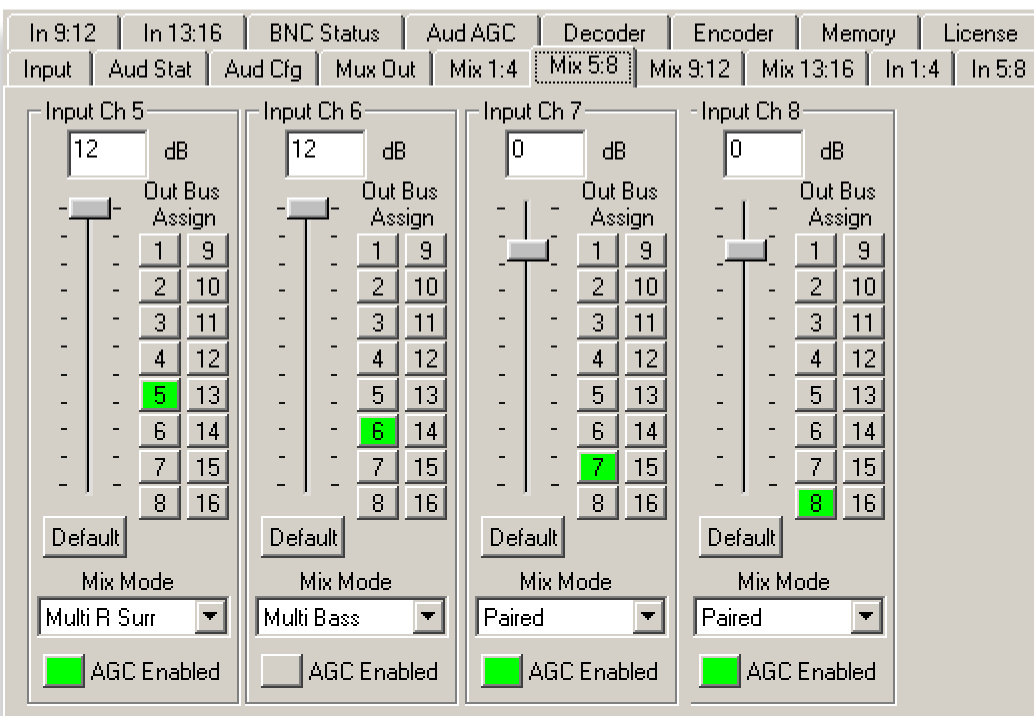
Mix Mode

For modules 7555, 7660, 7660-XV, 9550, 9550-XA, 9600 and 9600-XV, the **Mix Mode** drop-down control offers four possible selections for how to work with the channels: Normal, Paired, Surround Sound 5.1, and Surround Sound 7.1. Once you have established a pairing or Surround Sound grouping, changing the gain on one channel affects all of the associated channels.

AGC processing (if enabled) takes into account any channel pairs or Surround Sound groupings. These selections are described in the following table:

Mix Mode	Mixer Behavior
1. Normal	Working with mixer channels independently is the default or "Normal". mix mode.
2. Paired	If you want two channels to be paired so that altering the gain of one will automatically alter the gain of the other, choose Paired from the **Mix Mode** drop-down control for one of the channels you want to pair; for example, channel 9 and 10 will be paired with each other if you select Paired for one of those channels.
3. Surround Sound 5.1	For Surround Sound 5.1, which uses 6 channels, specify for each channel one of these 6 selections from the **Mix Mode** drop-down control. For example: Input Ch 1 = Multi Left Input Ch 2 = Multi Right Input Ch 3 = Multi Center Input Ch 4 = Multi L Surr Input Ch 5 = Multi R Surr Input Ch 6 = Multi Bass
4. Surround Sound 7.1	For Surround Sound 7.1, which uses 8 channels, specify for each channel one of the above 6 selections plus two additional Mix Mode selections. For example: Input Ch 7 = Multi L Rear Input Ch 8 = Multi R Rear





*Example of Avenue PC **Mix 1:4** and **Mix 5:8** Menus configured for Dolby Surround Sound 5.1 and a stereo pair.*

Out Bus Assignments

The mixer has 16 input channels and 16 output busses. Initially, each channel is assigned a separate output bus. For example, by default, mixer input channel 1 is assigned to mixer output bus 1, indicated by the green button in the **Input Ch 1** control. However, you can assign multiple input channels to go to the same output bus. Or you can have each input channel going to multiple output busses (from 0 to 16).

Input/Output Level Control

Each mixer channel has a level control on its input. There is not a separate output gain level control.

Configuring Audio Output

From the output of the mixer, you can send digital audio out through the 8 AES connectors. Analog audio is output through the 15-pin D connector. The digital and analog audio paths may be used simultaneously. You may also re-embed the audio.

Configuring Digital Audio Outputs

Use the **Out Bus Assign** controls from the **Mix 1:4**, **Mix 5:8**, **Mix 9:12** and **Mix 13:16** menus to route mixer inputs to mixer outputs. For digital audio, mixer output pair 1/2 feeds SDI out 1/2 and/or AES out 1/2. Mixer output pair 3/4 feeds SDI out 3/4 and/or AES out 3/4, and so on.

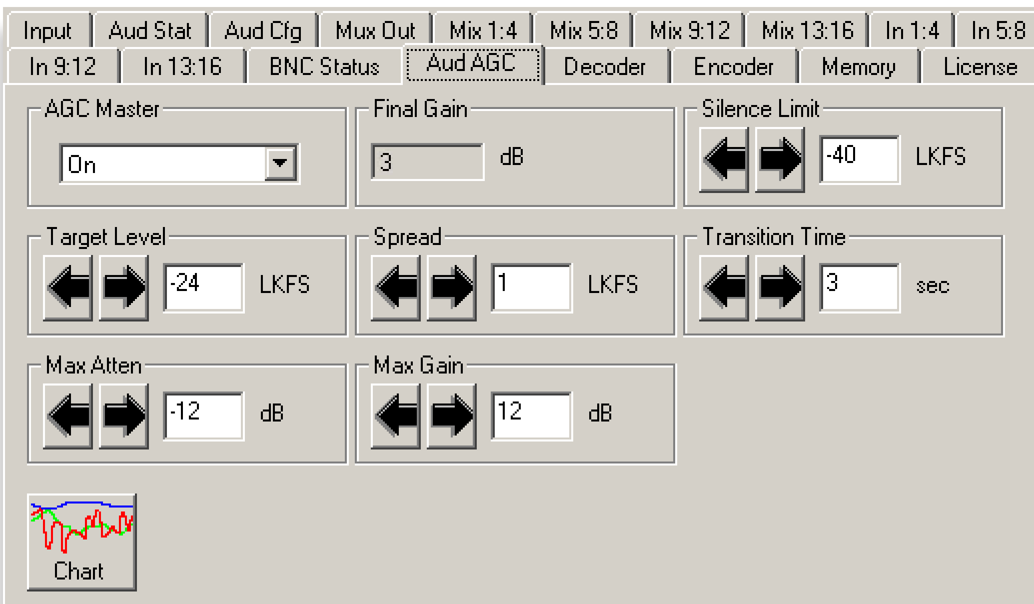
Note: If an AES connector is selected as an input, it cannot simultaneously be used as an output.

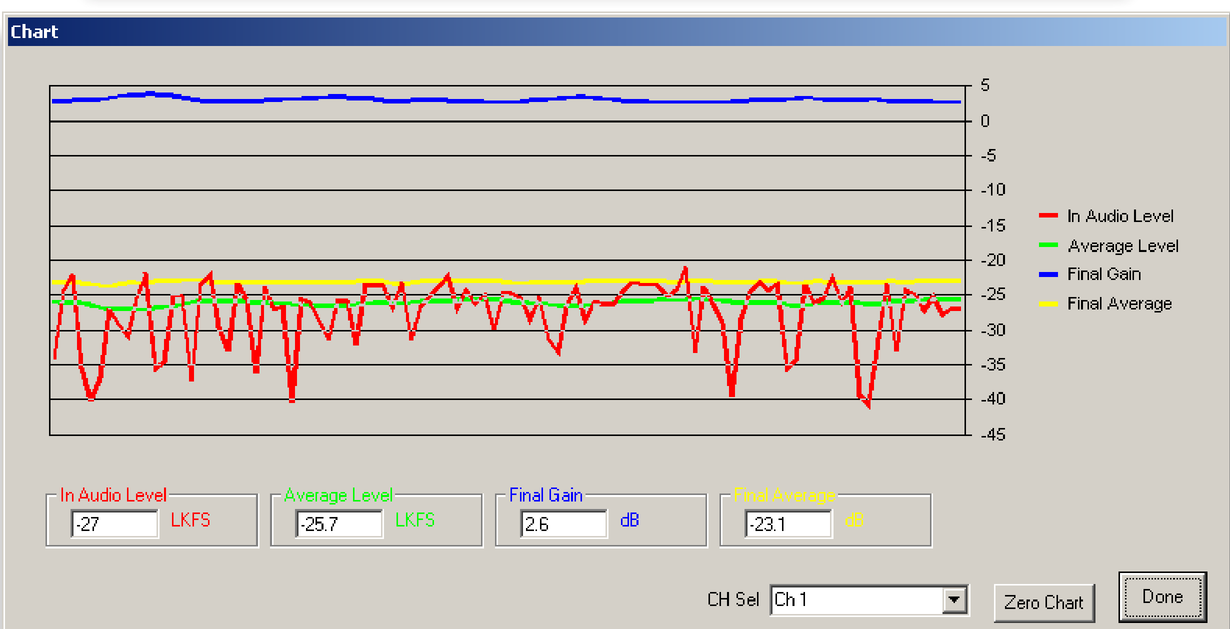
Configuration Example for Calm Act Compliance

Typical Settings

The 9670 LevelTrack Loudness Control Software is factory set to the typical settings used to comply with the Calm Act, as shown below. The chart shows an example of the AGC behavior with these settings.

AGC Master	On	Silence Limit	-40 LKFS
Target Level	-24 LKFS	Spread	1 LKFS
Transition Time	3 sec	Max Atten	-12 dB
Max Gain	12 LKFS		





Troubleshooting

The AGC is not being applied to the channel I am monitoring on the chart

For LevelTrack Audio Loudness Control AGC to function, the AGC Master must be turned on in the **Aud AGC** menu, and the channel(s) that you want AGC applied to must be enabled in the **Mix** menus.

To turn the AGC Master on, go to the **Aud AGC** menu and turn the AGC Master control On.

To enable the AGC on a channel by channel basis, go to the **Mix** menu applicable to the channel(s) that you want ACG applied to (for example, Channel 4 in Mix menu 1:4). Click the ACG Enable button at the bottom of the channel(s). The ACG Enable illuminates green when it is enabled and is grey when it is disabled. LevelTrack Audio Loudness Control AGC can be applied to all or any combination of the 16 audio channels.

The Mix Menus in Avenue PC are incomplete or garbled

Avenue PC Software version 2.0.15 or higher is required to fully support LevelTrack Loudness Control. Software updates are free at our website. You must have a valid serial number for Avenue PC in order to download the software.

To download the latest Avenue PC software, go to the following URL:

http://www.ensembledesigns.com/support/avenue-support/avenue-pc-software

The Mix Menus on the Touch Screen are not responding

At this time, the mixer menus (Mix 1:4, Mix 5:8, Mix 9:12, Mix 13:16) do not function with the Avenue Touch Screen interface. A pending software update will enable this control. However, all mixer functionality is currently available through the Avenue PC interface. Please be sure you have Avenue PC version 2.0.15 or higher installed.

Thirty-Day Demo

AGC provides functionality free for 30 days for demonstration purposes. After 30 days, it requires a valid key in order to continue working.

Warranty and Factory Service

Warranty

This module is covered by a five-year limited warranty. If you require service (under warranty or not), please contact Ensemble Designs and ask for customer service before you return the unit. This will allow the service technician an opportunity to provide any other suggestions for identifying the problem and to recommend possible solutions.

Factory Service

If you return equipment for repair, please get a Return Material Authorization Number (RMA) from the factory first.

tel +1 530.478.1830
fax +1 530.478.1832

service@ensembledesigns.com

www.ensembledesigns.com

Ship the product and a written description of the problem to:

Ensemble Designs, Inc.
Attention: Customer Service RMA #####
870 Gold Flat Rd.
Nevada City, CA 95959 USA

Be sure to put your RMA number on the outside of the box.

Ensemble White Paper on Dialnorm - Broadcaster Compliance with the Calm Act

In December of 2011, Congress enacted and the President signed the "Calm Act" regulating perceived loudness of programming at broadcast facilities in the U.S. This regulation -based on the ATSC A/85 Recommended Practice: Techniques for Establishing and Maintaining Audio Loudness for Digital Television – has caused a great deal of confusion in the market place. The key to understanding this legislation is the phrase "Perceived Loudness". Perceived loudness goes beyond simple VU gain values and AGC control. It also takes into account audio processing such as compression, expansion, gating, limiting, etc., that increases perceived loudness in program audio without increasing gain. The purpose of this legislation is to maintain consistent audio loudness to the consumer, both between channels and between programming and commercial content. Broadcasters, satellite providers, cable operators and other multi channel content providers have until December 2012 to comply. Ensemble Designs has products that can help broadcasters maintain compliance with the Calm Act.

Perceived loudness compliance is based on Dialog Normalization – dialnorm. Dialnorm is defined in ATSC A/85 as "An AC-3 metadata parameter, numerically equal to the absolute value of the Dialog Level, carried in the AC-3 bit stream. This unsigned 5-bit code indicates how far the average Dialog Level is below 0 LKFS. Valid values are 1-31. (zero value is reserved) The dialnorm values of 1 to 31 are interpreted as -1 to -31 LKFS."

To many, LKFS is a relatively new term, which means "Loudness K-weighted relative to Full Scale." It's a scale for audio measurement similar to VU or Peak. However, rather than measuring gain, it's measuring loudness. LKFS is based on the ITU-R BS.1770 Loudness Measurement Method. The ITU-R group performed tests to groups of people around the world with various content and listening environments and were able to construct an algorithm that accurately measured the loudness of audio content. It was quite an undertaking, and the details of the process take up the majority of the BS.1770 document. Have a look sometime. It covers everything from speaker placement to noise in the environment, to the type of content used for the testing. The loudness unit of LKFS is dB and is used the same way as a dB of gain. For example, a -15 LKFS program can be made to match the loudness of a -22 LKFS program by attenuating 7 dB.

There is a defined process for determining the dialnorm value for a particular piece of content. It's a fairly involved process depending on whether dialog is present and the length of the sample that must be used to determine the dialnorm value for the entire program. It can be found in the ATSC A/85 document if you'd like more detail. Dialnorm relates to the level of an "Anchor Element" which is usually dialog. It disregards momentary intentional loud elements such as gunshots or car crashes. Here's another way to think about this. When you are home watching TV, what do you adjust the volume to hear? It's usually the talking part of the program. Sound effects may be louder, whispering softer. But when you adjust the TV volume, you're setting it to comfortably hear what's being spoken. This is essentially dialnorm. If there is no dialog present, as in music programming, it relates to the viewer focus element. For example: the level of the featured pianist rather than the level of the entire orchestra. A single dialnorm value is determined for the entire program content, and is embedded into the metadata bit stream of the SDI signal. To simplify the process, companies such as Dolby, Sony, Tektronix and others, have a box that will "listen" to the program content and then report a dialnorm value for that piece of programming. Another box is often used to encode this value into the SDI stream. File based content solutions are also available from companies such as Telestream and Masstech which analyze a particular file's content and then embeds a dialnorm into the metadata of the file. All this is well and good for content that is already produced, but what about live productions such as sports or news? For this type of programming, a dialnorm target is selected. This is usually specified by network specs or distributor standard practices. The audio technician then mixes the audio content to the target dialnorm using LKFS metering. The dialnorm metadata (value) is entered and encoded into the SDI stream on the fly.

The AC-3 audio system defined in the ATSC Digital Television Standard uses dialnorm metadata to control loudness and other audio parameters more effectively without permanently altering the dynamic range of the content. The content provider or DTV operator encodes metadata (the dialnorm value) along with the audio content into an AC-3 encoder. This metadata parameter, when extracted at the decoder, sets different content to a uniform loudness transparently. Basically, it provides results similar to the viewer using his remote control to set a comfortable volume between disparate TV programs, commercials and channels.

So, what does this mean to the average content distributor who has to comply with the Calm Act? The FCC is attempting to regulate differences in perceived loudness with particular attention being paid to inter-channel audio levels; the difference between programming content and commercial content. In an ideal world, all content providers would provide accurate dialnorm values in the audio bitstream of their programming. These values, sent to the AC-3 encoder before transmission would then be decoded at the consumer set top box or flat screen display, which automatically adjusts "volume" levels at the receiver based on the values received in the metadata. This works by Dynamic Range Control "gain words" calculated in the encoder and then applied at the decoder. These DRC calculations are relative to and based on the indicated loudness of content represented by the dialnorm metadata parameter. In other words, the encoder needs to know how loud the content is intended to be so it can determine when the content is either "too loud" or "too quiet". Dialnorm effectively sets this target. Therefore, it's very important that the dialnorm accurately indicates the loudness of the content.

The concept of Fixed metadata is simply to "fix" the AC-3 encoder dialnorm setting to a single value and to bring the loudness of the encoder audio input signal into conformance with this setting. This is the simplest method with no requirement for additional metadata equipment or data management. It is the only approach possible when using an encoder without metadata input or external GPI control.

The Preset metadata concept uses GPI triggers to set predetermined preset values to be loaded into the AC-3 encoder to accommodate known differences in content loudness. For example, known differences between network feed and local playout. Some AC-3 encoders however, reset and disrupt the audio bit stream output when a preset is changed. Depending on the encoder, this may result in an audible "glitch" on air. To avoid this potential problem, it may be necessary to provide a framesync for the output of the AC-3 encoder to stabilize the AC-3 source.

The Agile metadata system allows setting different dialnorm values for different content that has different loudness. This is accomplished by embedding the dialnorm parameter within the metadata bit stream accompanying the content at an upstream location. The metadata is dis-embedded just prior to the AC-3 encoder and then connected to the encoder's external serial metadata input. The encoder dialnorm setting then changes appropriately on the boundaries of the content. The downside of the Agile metadata system is the potential for a severe discrepancy in loudness between programs and between stations if metadata is lost. Encoders with external metadata input provide a "reversion" feature to mitigate the impact of metadata loss. It can be configured to either retain the most recent metadata value, or revert to an operator-defined preset. While this feature can minimize the impact on the consumer, the error in loudness can still be significant. This method requires each piece of content submitted for broadcast to have its unique dialnorm value embedded into the audio bit stream.

In the real world, content provided to a broadcaster doesn't always contain a valid dialnorm value. Much of the commercial content received at the local level contains none whatsoever. In such cases, the target loudness value should be -24 LKFS (+/- 2dB). This equates to a dialnorm value of 24. Broadcasters should be using a BS.1770 metering system to determine proper LKFS values, and all content received needs to have the dialnorm embedded prior to AC-3 encoding.

Until the "ideal world" becomes a reality, it may be necessary to have a device that maintains audio levels at a particular dialnorm value, particularly when using a fixed dialnorm metadata AC-3 encoder. Enter the Ensemble Designs 9670 LevelTrack software key for the Avenue 9600, 9550, 7660 and 7555. This software uses the BS.1770 loudness algorithm to set a specific LKFS value to be maintained by the audio stream. As mentioned earlier, this translates directly to a dialnorm value.

Station output audio that is run through one of the Ensemble Designs products with the 9670 software key enabled, can be preset to a specific LKFS value before hitting the fixed dialnorm metadata EC-3 encoder. As an example, if the encoder has been set for a fixed dialnorm of 24, the 9670 software would be set to -24 LKFS – in effect feeding the encoder an audio loudness dialnorm equivalent of 24. This allows the broadcaster to use a lower cost encoder and still maintain consistent loudness levels required by the Calm Act.

The FCC mentions that enforcement of the Calm Act will be complaint driven. If stations show a consistent pattern of complaints related to audio level disparity, the FCC will investigate. This is where the Ensemble Designs 9690 Audio Compliance and Monitoring software along with any of the products mentioned earlier will provide a record of LKFS levels of up to four devices. This record can be used to prove compliance.

By installing and properly setting up a 7555, 7660, 9550 or 9600 with 9670 software key for LKFS AGC, and 9690 software key for compliance recording, broadcasters can rest assured that they are in compliance with the Calm Act, and limit audio loudness complaints by their viewers. That's a winwin.

Calm Act Update – NAB 2012: The word around NAB this year is that the ATSC A/85 committee will reconvene shortly to make modifications to the A/85 Recommended Practice for Maintaining Audio Loudness upon which the Calm Act is based.

Specifically, the committee is meeting to discuss standards for the use of "gating" in LKFS averaging. This process removes audio below a certain threshold, from the LKFS averaging equation. They will be meeting to determine exactly where and how the threshold will be implemented.

The problem with the existing spec is that silence is included in the LKFS averaging to determine dialnorm. This means that if a broadcaster were to air a 30 second spot that is silent – (has no audio), they are in effect, non-compliant with the CALM Act (+/- 2 dB from average). In addition, the silence will effect the averaging after audio resumes causing a louder than normal perceived loudness until proper averaging returns things to normal.

The committee will address these issues. It has an implication to broadcasters and manufacturers alike. LKFS metering will have to take into account the threshold of the gating before determining overall LKFS average. File based dialnorm solutions will have to rework their algorithms. Broadcasters of course, will still have to be in compliance with the Calm Act. All of this has to be done by the December of this year. Ensemble Designs will be keeping a close eye on these proceedings and will be looking to make enhancements to our AGC and Compliance software to reflect changes to the spec. Stay tuned.

Glossary

AES/EBU

The digital audio standard defined as a joint effort of the Audio Engineering Society and the European Broadcast Union. AES/EBU or AES3 describes a serial bitstream that carries two audio channels, thus an AES stream is a stereo pair. The AES/EBU standard covers a wide range of sample rates and quantizations (bit depths). In television systems, these will generally be 48 KHz and either 20 or 24 bits.

AFD

Active Format Description is a method to carry information regarding the aspect ratio of the video content. The specification of AFD was standardized by SMPTE in 2007 and is now beginning to appear in the marketplace. AFD can be included in both SD and HD SDI transport systems. There is no legacy analog implementation. (See WSS).

ASI

A commonly used transport method for MPEG video streams, ASI or Asynchronous Serial Interface, operates at the same 270 Mb/s data rate as SD SDI. This makes it easy to carry an ASI stream through existing digital television infrastructure. Known more formally as DVB-ASI, this transport mechanism can be used to carry multiple program channels.

Aspect Ratio

The ratio of the vertical and horizontal measurements of an image. 4:3 is the aspect ratio for standard definition video formats and television and 16:9 for high definition. Converting formats of unequal ratios is done by letterboxing (horizontal bars) or pillar boxing (vertical pillars) in order to keep the original format's aspect ratio.

Bandwidth

Strictly speaking, this refers to the range of frequencies (i.e. the width of the band of frequency) used by a signal, or carried by a transmission channel. Generally, wider bandwidth will carry and reproduce a signal with greater fidelity and accuracy.

Beta

Sony Beta SP video tape machines use an analog component format that is similar to SMPTE, but differs in the amplitude of the color difference signals. It may also carry setup on the luminance channel.

Bit

A binary digit, or bit, is the smallest amount of information that can be stored or transmitted digitally by electrical, optical, magnetic, or other means. A single bit can take on one of two states: On/Off, Low/High, Asserted/ Deasserted, etc. It is represented numerically by the numerals 1 (one) and 0 (zero). A byte, containing 8 bits, can represent 256 different states. The binary number 11010111, for example, has the value of 215 in our base 10 numbering system. When a value is carried digitally, each additional bit of resolution will double the number of different states that can be represented. Systems that operate with a greater number of bits of resolution, or quantization, will be able to capture a signal with more detail or fidelity. Thus, a video digitizer with 12 bits of resolution will capture 4 times as much detail as one with 10 bits.

Blanking

The Horizontal and Vertical blanking intervals of a television signal refer to the time periods between lines and between fields. No picture information is transmitted during these times, which are required in CRT displays to allow the electron beam to be repositioned for the start of the next line or field. They are also used to carry synchronizing pulses which are used in transmission and recovery of the image. Although some of these needs are disappearing, the intervals themselves are retained for compatibility purposes. They have turned out to be very useful for the transmission of additional content, such as teletext and embedded audio.

CAV

Component Analog Video. This is a convenient shorthand form, but it is subject to confusion. It is sometimes used to mean ONLY color difference component formats (SMPTE or Beta), and other times to include RGB format. In any case, a CAV signal will always require 3 connectors – either Y/R-Y/B-Y, or R/G/B.

Checkfield

A Checkfield signal is a special test signal that stresses particular aspects of serial digital transmission. The performance of the Phase Locked-Loops (PLLs) in an SDI receiver must be able to tolerate long runs of 0's and 1's. Under normal conditions, only very short runs of these are produced due to a scrambling algorithm that is used. The Checkfield, also referred to as the Pathological test signal, will "undo" the scrambling and cause extremely long runs to occur. This test signal is very useful for testing transmission paths.

Chroma

The color or chroma content of a signal, consisting of the hue and saturation of the image.
See also Color Difference.

Component

In a component video system, the totality of the image is carried by three separate but related components. This method provides the best image fidelity with the fewest artifacts, but it requires three independent transmission paths (cables). The commonly used component formats are Luminance and Color Difference (Y/Pr/Pb), and RGB. It was far too unwieldy in the early days of color television to even consider component transmission.

Composite

Composite television dates back to the early days of color transmission. This scheme encodes the color difference information onto a color subcarrier. The instantaneous phase of the subcarrier is the color's hue, and the amplitude is the color's saturation or intensity. This subcarrier is then added onto the existing luminance video signal. This trick works because the subcarrier is set at a high enough frequency to leave spectrum for the luminance information. But it is not a seamless matter to pull the signal apart again at the destination in order to display it or process it. The resultant artifacts of dot crawl (also referred to as chroma crawl) are only the most obvious result. Composite television is the most commonly used format throughout the world, either as PAL or NTSC. It is also referred to as Encoded video.

Color Difference

Color Difference systems take advantage of the details of human vision. We have more acuity in our black and white vision than we do in color. This means that we need only the luminance information to be carried at full bandwidth, we can scrimp on the color channels. In order to do this, RGB information is converted to carry all of the luminance (Y is the black and white of the scene) in a single channel. The other two channels are used to carry the "color difference". Noted as B-Y and R-Y, these two signals describe how a particular pixel "differs" from being purely black and white. These channels typically have only half the bandwidth of the luminance.

Decibel (dB)

The decibel is a unit of measure used to express the ratio in the amplitude or power of two signals. A difference of 20 dB corresponds to a 10:1 ratio between two signals, 6 dB is approximately a 2:1 ration. Decibels add while the ratios multiply, so 26 dB is a 20:1 ratio, and 14 dB is a 5:1 ratio. There are several special cases of the dB scale, where the reference is implied. Thus, dBm refers to power relative to 1 milliwatt, and dBu refers to voltage relative to .775V RMS. The original unit of measure was the Bel (10 times bigger), named after Alexander Graham Bell.

dBFS

In Digital Audio systems, the largest numerical value that can be represented is referred to as Full Scale. No values or audio levels greater than FS can be reproduced because they would be clipped. The nominal operating point (roughly corresponding to 0 VU) must be set below FS in order to have headroom for audio peaks. This operating point is described relative to FS, so a digital reference level of -20 dBFS has 20 dB of headroom before hitting the FS clipping point.

DVI

Digital Visual Interface. DVI-I (integrated) provides both digital and analog connectivity. The larger group of pins on the connector are digital while the four pins on the right are analog.

EDH

Error Detection and Handling is a method to verify proper reception of an SDI or HD-SDI signal at the destination. The originating device inserts a data packet in the vertical interval of the SDI signal and every line of the HD signal which contains a checksum of the entire video frame. This checksum is formed by adding up the numerical values of all of the samples in the frame, using a complex formula. At the destination this same formula is applied to the incoming video and the resulting value is compared to the one included in the transmission. If they match, then the content has all arrived with no errors. If they don't, then an error has occurred.

Embedded Audio

Digital Audio can be carried along in the same bitstream as an SDI or HD-SDI signal by taking advantage of the gaps in the transmission which correspond to the horizontal and vertical intervals of the television waveform. This technique can be very cost effective in transmission and routing, but can also add complexity to signal handling issues because the audio content can no longer be treated independently of the video.

Eye Pattern

To analyze a digital bitstream, the signal can be displayed visually on an oscilloscope by triggering the horizontal timebase with a clock extracted from the stream. Since the bit positions in the stream form a very regular cadence, the resulting display will look like an eye – an oval with slightly pointed left and right ends. It is easy to see from this display if the eye is "open", with a large central area that is free of negative or positive transitions, or "closed" where those transitions are encroaching toward the center. In the first case, the open eye indicates that recovery of data from the stream can be made reliably and with few errors. But in the closed case data will be difficult to extract and bit errors will occur. Generally it is jitter in the signal that is the enemy of the eye.

Frame Sync

A Frame Synchronizer is used to synchronize the timing of a video signal to coincide with a timing reference (usually a color black signal that is distributed throughout a facility). The synchronizer accomplishes this by writing the incoming video into a frame buffer memory under the timing direction of the sync information contained in that video. Simultaneously the memory is being read back by a timing system that is genlocked to a house reference. As a result, the timing or alignment of the video frame can be adjusted so that the scan of the upper left corner of the image is happening simultaneously on all sources. This is a requirement for both analog and digital systems in order to perform video effects or switch glitch-free in a router. Frame synchronization can only be performed within a single television line standard. A synchronizer will not convert an NTSC signal to a PAL signal, it takes a standards converter to do that.

Frequency Response

A measurement of the accuracy of a system to carry or reproduce a range of signal frequencies. Similar to Bandwidth.

H.264

The latest salvo in the compression wars is H.264 which is also known as MPEG-4 Part 10. MPEG-4 promises good results at just half the bit rate required by MPEG-2.

HD

High Definition. This two letter acronym has certainly become very popular. Here we thought it was all about the pictures – and the radio industry stole it.

HDCP

(High-bandwidth Digital Content Protection) is a content encryption system for HDMI. It is meant to prevent copyright content from being copied. Protected content, like a movie on a Blu-Ray disc is encrypted by its creator. Devices that want to display the protected content, like a television, must have an authorized key in order to decode the signal and display it. The entity that controls the HDCP standard strictly limits the kinds of devices that are allowed decryption keys. Devices that decrypt the content and provide an unencrypted copy are not allowed.

HDMI

The High Definition Multimedia Interface comes to us from the consumer marketplace where it is becoming the de facto standard for the digital interconnect of display devices to audio and video sources. It is an uncompressed, all-digital interface that transmits digital video and eight channels of digital audio. HDMI is a bit serial interface that carries the video content in digital component form over multiple twisted-pairs. HDMI is closely related to the DVI interface for desktop computers and their displays.

IEC

The International Electrotechnical Commission provides a wide range of worldwide standards. They have provided standardization of the AC power connection to products by means of an IEC line cord. The connection point uses three flat contact blades in a triangular arrangement, set in a rectangular connector. The IEC specification does not dictate line voltage or frequency. Therefore, the user must take care to verify that a device either has a universal input (capable of 90 to 230 volts, either 50 or 60 Hz), or that a line voltage switch, if present, is set correctly.

Interlace

Human vision can be fooled to see motion by presenting a series of images, each with a small change relative to the previous image. In order to eliminate the flicker, our eyes need to see more than 30 images per second. This is accomplished in television systems by dividing the lines that make up each video frame (which run at 25 or 30 frames per second) into two fields. All of the odd-numbered lines are transmitted in the first field, the even-numbered lines are in the second field. In this way, the repetition rate is 50 or 60 Hz, without using more bandwidth. This trick has worked well for years, but it introduces other temporal artifacts. Motion pictures use a slightly different technique to raise the repetition rate from the original 24 frames that make up each second of film—they just project each one twice.

IRE

Video level is measured on the IRE scale, where 0 IRE is black, and 100 IRE is full white. The actual voltages that these levels correspond to can vary between formats.

ITU-R 601

This is the principal standard for standard definition component digital video. It defines the luminance and color difference coding system that is also referred to as 4:2:2. The standard applies to both PAL and NTSC derived signals. They both will result in an image that contains 720 pixels horizontally, with 486 vertical pixels in NTSC, and 576 vertically in PAL. Both systems use a sample clock rate of 27 MHz, and are serialized at 270 Mb/s.

Jitter

Serial digital signals (either video or audio) are subject to the effects of jitter. This refers to the instantaneous error that can occur from one bit to the next in the exact position of each digital transition. Although the signal may be at the correct frequency on average, in the interim it varies. Some bits come slightly early, others come slightly late. The measurement of this jitter is given either as the amount of time uncertainty or as the fraction of a bit width. For 270 Mb/s SD video, the allowable jitter is 740 picoseconds, or 0.2 UI (Unit Interval – one bit width). For 1.485 Gb/s HD, the same 0.2UI spec corresponds to just 135 pico seconds.

Luminance

The "black & white" content of the image. Human vision had more acuity in luminance, so television systems generally devote more bandwidth to the luminance content. In component systems, the luminance is referred to as Y.

MPEG

The Moving Picture Experts Group is an industry group that develops standards for the compression of moving pictures for television. Their work is an on-going effort. The understanding of image processing and information theory is constantly expanding. And the raw bandwidth of both the hardware and software used for this work is ever increasing. Accordingly, the compression methods available today are far superior to the algorithms that originally made the real-time compression and decompression of television possible. Today, there are many variations of these techniques, and the term MPEG has to some extent become a broad generic label.

Metadata

This word comes from the Greek, meta means 'beyond' or 'after'. When used as a prefix to 'data', it can be thought of as 'data about the data'. In other words, the metadata in a data stream tells you about that data – but it is not the data itself. In the television industry, this word is sometimes used correctly when, for example, we label as metadata the timecode which accompanies a video signal. That timecode tells you something about the video, i.e. when it was shot, but the timecode in and of itself is of no interest. But in our industry's usual slovenly way in matters linguistic, the term metadata has also come to be used to describe data that is associated with the primary video in a datastream. So embedded audio will (incorrectly) be called metadata when it tells us nothing at all about the pictures.

Multi-mode

Multi-mode fibers have a larger diameter core than single mode fibers (either 50 or 62.5 microns compared to 9 microns), and a correspondingly larger aperture. It is much easier to couple light energy into a multi-mode fiber, but internal reflections will cause multiple "modes" of the signal to propagate down the fiber. This will degrade the ability of the fiber to be used over long distances.
See also Single Mode.

NTSC

The color television encoding system used in North America was originally defined by the National Television Standards Committee. This American standard has also been adopted by Canada, Mexico, Japan, Korea, and Taiwan. (This standard is referred to disparagingly as Never Twice Same Color.)

Optical

An optical interface between two devices carries data by modulating a light source. This light source is typically a laser or laser diode (similar to an LED) which is turned on and off at the bitrate of the datastream. The light is carried from one device to another through a glass fiber. The fiber's core acts as a waveguide or lightpipe to carry the light energy from one end to another. Optical transmission has two very significant advantages over metallic copper cables. Firstly, it does not require that the two endpoint devices have any electrical connection to each other. This can be very advantageous in large facilities where problems with ground loops appear. And secondly, and most importantly, an optical interface can carry a signal for many kilometers or miles without any degradation or loss in the recovered signal. Copper is barely useful at distances of just 1000 feet.

Oversampling

A technique to perform digital sampling at a multiple of the required sample rate. This has the advantage of raising the Nyquist Rate (the maximum frequency which can be reproduced by a given sample rate) much higher than the desired passband. This allows more easily realized anti-aliasing filters.

PAL

During the early days of color television in North America, European broadcasters developed a competing system called Phase Alternation by Line. This slightly more complex system is better able to withstand the differential gain and phase errors that appear in amplifiers and transmission systems. Engineers at the BBC claim that it stands for Perfection At Last.

Pathological Test Pattern – see Checkfield

Progressive

An image scanning technique which progresses through all of the lines in a frame in a single pass. Computer monitors all use progressive displays. This contrasts to the interlace technique common to television systems.

Return Loss

An idealized input or output circuit will exactly match its desired impedance (generally 75 ohms) as a purely resistive element, with no reactive (capacitive or inductive) elements. In the real world, we can only approach the ideal. So, our real inputs and outputswill have some capacitance and inductance. This will create impedance matching errors, especially at higher frequencies. The Return Loss of an input or output measures how much energy is returned (reflected back due to the impedance mismatch). For digital circuits, a return loss of 15 dB is typical. This means that the energy returned is 15 dB less than the original signal. In analog circuits, a 40 dB figure is expected.

RGB

RGB systems carry the totality of the picture information as independent Red, Green, and Blue signals. Television is an additive color system, where all three components add to produce white. Because the luminance (or detail) information is carried partially in each of the RGB channels, all three must be carried at full bandwidth in order to faithfully reproduce an image.

ScH Phase

Used in composite systems, ScH Phase measures the relative phase between the leading edge of sync on line 1 of field 1 and a continuous subcarrier sinewave. Due to the arithmetic details of both PAL and NTSC, this relationship is not the same at the beginning of each frame. In PAL, the pattern repeats ever 4 frames (8 fields) which is also known as the Bruch Blanking sequence. In NTSC, the repeat is every 2 frames (4 fields). This creates enormous headaches in editing systems and the system timing of analog composite facilities.

Setup

In the NTSC Analog Composite standard, the term Setup refers to the addition of an artificial offset or pedestal to the luminance content. This places the Black Level of the analog signal 54 mV (7.5 IRE) positive with respect to ground. The use of Setup is a legacy from the early development of television receivers in the vacuum tube era. This positive offset helped to prevent the horizontal retrace of the electron beam from being visible on the CRT, even if Brightness and Contrast were mis-adjusted. While the use of Setup did help to prevent retrace artifacts, it did so at the expense of dynamic range (contrast) in the signal because the White Level of the signal was not changed.

Setup is optional in NTSC systems, but is never used in PAL systems (see 'Perfection' characteristic of PAL). This legacy of Setup continues to persist in North American NTSC systems, while it has been abandoned in Japan.

In the digital component world (SD and HD SDI) there is obviously no need for, and certainly every reason to avoid, Setup. In order for the interfaces between analog and digital systems to operate as transparently as possible, Setup must be carefully accounted for in conversion products. When performing analog to digital conversion, Setup (if present) must be removed and the signal range gained up to account for the 7.5% reduction in dynamic range. And when a digital signal is converted back to analog form, Setup (if desired on the output) must be created by reducing the dynamic range by 7.5% and adding the 54 mV positive offset. Unfortunately, there is no truly foolproof algorithm to detect the presence of Setup automatically, so it's definitely a case of installer beware.

SDI

Serial Digital Interface. This term refers to inputs and outputs of devices that support serial digital component video. This could refer to standard definition at 270 Mb/s, HD SDI or High Definition Serial Digital video at 1.485 Gb/s, or to the newer 3G standard of High Definition video at 2.97 Gb/s.

SMPTE

The Society of Motion Picture and Television Engineers is a professional organization which has done tremendous work in setting standards for both the film and television industries. The term "SMPTE"' is also shorthand for one particular component video format - luminance and color difference.

Single Mode

A Single mode (or mono mode) optical fiber carries an optical signal on a very small diameter (9 micron) core surrounded with cladding. The small diameter means that no internally reflected lightwaves will be propagated. Thus only the original "mode" of the signal passes down the fiber. A single mode fiber used in an optical SDI system can carry a signal for up to 20 kilometers. Single mode fibers require particular care in their installation due to the extremely small optical aperture that they present at splice and connection points. See also Multi-mode.

TBC

A Time Base Corrector is a system to reduce the Time Base Error in a signal to acceptable levels. It accomplishes this by using a FIFO (First In, First Out) memory. The incoming video is written into the memory using its own jittery timing. This operation is closely associated with the actual digitization of the analog signal because the varying position of the sync timing must be mimicked by the sampling function of the analog to digital converter. A second timing system, genlocked to a stable reference, is used to read the video back out of the memory. The memory acts as a dynamically adjusting delay to smooth out the imperfections in the original signal's timing. Very often a TBC will also function as a Frame Synchronizer. See also Frame Sync.

Time Base Error

Time base error is present when there is excessive jitter or uncertainty in the line to line output timing of a video signal. This is commonly associated with playback from video tape recorders, and is particularly severe with consumer type heterodyne systems like VHS. Time base error will render a signal unusable for broadcast or editing purposes.

Timecode

Timecode, a method to uniquely identify and label every frame in a video stream, has become one of the most recognized standards ever developed by SMPTE. It uses a 24 hour clock, consisting of hours, minutes, seconds, and television frames. Originally recorded on a spare audio track, this 2400 baud signal was a significant contributor to the development of video tape editing. We now refer to this as LTC or Longitudinal Time Code because it was carried along the edge of the tape. This allowed it to be recovered in rewind and fast forward when the picture itself could not. Timecode continues to be useful today and is carried in the vertical interval as VITC, and as a digital packet as DVITC. Timecode is the true metadata.

Tri-Level Sync

For many, many years, television systems used composite black as a genlock reference source. This was a natural evolution from analog systems to digital implementations. With the advent of High Definition television, with even higher data rates and tighter jitter requirements, problems with this legacy genlock signal surfaced. Further, a reference signal with a 50 or 60 Hz frame rate was useless with 24 Hz HD systems running at film rates. Today we can think of composite black as a bi-level sync signal – it has two levels, one at sync tip and one at blanking. For HD systems, Tri-Level Sync, which has the same blanking level (at ground) of bi-level sync, but the sync pulse now has both a negative and a positive element. This keeps the signal symmetrically balanced so that its DC content is zero. And it also means that the timing pickoff point is now at the point where the signal crosses blanking and is no longer subject to variation with amplitude. This makes Tri-Level Sync a much more robust signal and one which can be delivered with less jitter.

USB

The Universal Serial Bus, developed in the computer industry to replace the previously ubiquitous RS-232 serial interface, now appears in many different forms and with many different uses. It actually forms a small local area network, allowing multiple devices to coexist on a single bus where they can be individually addressed and accessed.

VGA

Video Graphics Array. Traditional 15-pin, analog interface between a PC and monitor.

Word Clock

Use of Word Clock to genlock digital audio devices developed in the audio recording industry. Early digital audio products were interconnected with a massive parallel connector carrying a twisted pair for every bit in the digital audio word. A clock signal, which is a square wave at the audio sampling frequency, is carried on a 75 ohm coaxial cable. Early systems would daisychain this 44.1 or 48 kilohertz clock from one device to another with coax cable and Tee connectors. On the rising edge of this Work Clock these twisted pairs would carry the left channel, while on the falling edge, they would carry the right channel. In most television systems using digital audio, the audio sample clock frequency (and hence the 'genlock' between the audio and video worlds) is derived from the video genlock signal. But products that are purely audio, with no video reference capability, may still require Word Clock.

WSS

Wide Screen Signaling is used in the PAL/625 video standards, both in analog and digital form, to convey information about the aspect ratio and format of the transmitted signal. Carried in the vertical interval, much like closed captioning, it can be used to signal a television receiver to adjust its vertical or horizontal sizing to reflect incoming material. Although an NTSC specification for WSS exists, it never achieved any traction in the marketplace.

YUV

Strictly speaking, YUV does not apply to component video. The letters refer to the Luminance (Y), and the U and V encoding axes using in the PAL composite system. Since the U axis is very close to the B-Y axis, and the V axis is very close to the R-Y axis, YUV is often used as a sort of shorthand for the more long-winded "Y/R-Y/B-Y".

Y/Cr/Cb

In digital component video, the luminance component is Y, and the two color difference signals are Cr (R-Y) and Cb (B-Y).

Y/Pr/Pb

In analog component video, the image is carried in three components. The luminance is Y, the R-Y color difference signal is Pr, and the B-Y color difference signal is Pb.