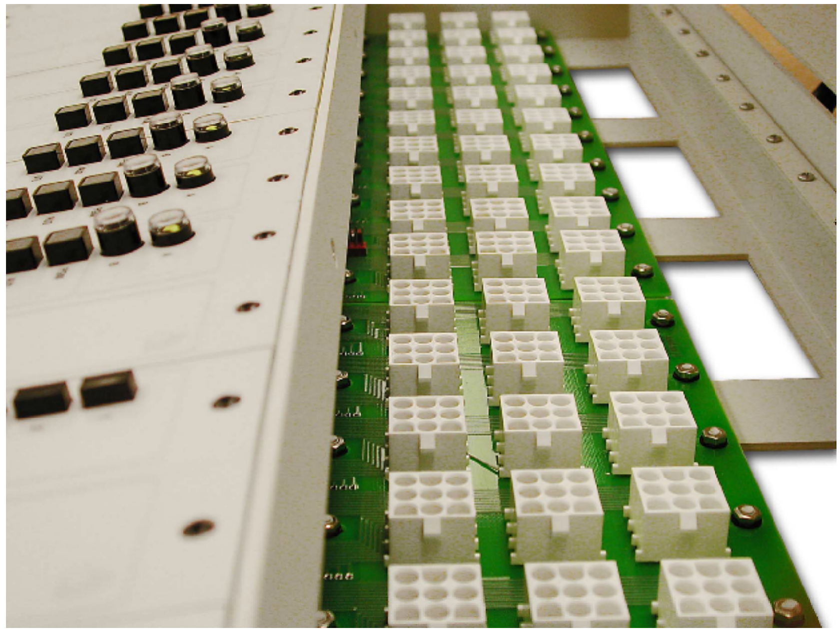
Audio Levels

Arrakis Systems inc.

application note



Purpose of this Ap Note

This application note is designed as a practical aid for designing, installing, and debugging low noise, high performance audio broadcast studios and facilities. It is intended for use by novice and experienced "technical" people alike, including managers.

The application note focuses on the basic principles of audio "systems" design. Simple mathematical models are used only as they illustrate a principle. We find that it is the proper understanding and application of basic principles that results in a professional audio installation. It is often only through an application of basic principles that a problematic installation can be corrected.

In preparation for writing this application note, we have performed an extensive review of available technical literature and product manuals on these subjects. The review underlined the complexity of modern audio systems design and that this is a field under constant change. Combining audio products from the broadcast, consumer, music, commercial sound, and now personal computer industries into a single facility is a challenge. These different industries have different product design goals that have resulted in an inability to simply "plug and play." It would be thought that it would be possible to simply purchase equipment and off the shelf interconnection cables to assemble an audio facility. However, variations in audio levels, impedance, connector designs, AC and audio ground systems, and other factors make this difficult. The purpose of this application note is to help to provide enough of an understanding of the underlying principles to be able to overcome these obstacles.

Arrakis Systems has been building professional radio consoles since the late 1970's and digital audio source equipment since the early 1990's. We are a leading manufacturer and innovator in the professional broadcast audio industry. We have accumulated experience with thousands of studios in diverse conditions around the world.

I- Important Information

Important Disclaimer

Use of the information in this document is entirely at the reader's risk. Although precaution has been taken in the preparation of this document, Arrakis Systems inc. assumes no responsibility for errors or omissions. Neither is any liability assumed for damages resulting from the use of the information contained herein.

Danger- Shock & other hazards

Electronic products may contain potentially lethal voltages and currents and should be serviced by trained and experienced personnel only. Any installation, test, or calibration procedures in this document that require access to the interior of the equipment should be performed by qualified personnel only.

How to Contact Arrakis

Arrakis Systems inc. is located at		Arrakis Systems inc 6604 Powell Street Loveland, Colorado 80538
Business Hours:		8:00am - 4:30pm mountain time
Contact:	Voice:	970-461-0730
	Fax:	970-663-1010
	email:	support@arrakis-systems.com
	web:	www.arrakis-systems.com

Having difficulty contacting Arrakis? Refer to the website (www.arrakis-systems.com) for current contact information

ANALOG & DIGITAL LEVELS

15.0 Audio Levels

Audio systems have three important signal levels. These are:

(1) Maximum level-	if the level exceeds this the signal is heavily distorted
(2) Nominal level-	the average audio level as measured as 0VU on the meters
(3) Noise floor-	the minimum signal level in the system

a) ANALOG MAXIMUM LEVEL

In an analog system, the maximum level is typically the point where the signal reaches or exceeds the maximum level that any audio stage in the system can produce. This can appear at the input stage of a product or signal chain, the output stage, or anywhere in between. This is typically the limit set by the supply voltages of the equipment. An amplifier with a (+) (-) 12 VDC supply can not produce a signal with a peak to peak swing of more than 24 volts. This is called “clipping” because the top of an audio signal is “clipped” off when the signal reaches this level. Clipping in an analog signal chain is progressive as the signal increases until it approaches a square wave.

TYPICAL MAXIMUM OUTPUTS

(+)(-)12VDC	balanced	+22dBu
(+)(-)15VDC	balanced	+26dBu
(+)(-)12VDC	unbalanced	+16dBu

b) ANALOG NOMINAL LEVEL

In professional equipment,	the 0VU level = +4dBu	(1.228 VRMS)
In consumer equipment,	the 0VU level = 0dBV	(316 millivolts RMS)

c) HEADROOM

Headroom is the number of dB that the signal can be increased about the nominal level (0VU) before the maximum signal (clipping) is reached.

Therefore:	Headroom = Maximum Signal (dB) - 0VU (dB)
An example would be:	Headroom = +22dBu - (+)4dBu = 18dB.

Headroom is an important defining specification for any audio system because any signal peak above it will be heavily distorted.

c) VU METERS & PEAK METERS

VU meters are average responding meters. If a VU meter is reading 0VU with audio program material then the actual audio peaks are many dB higher and not visible on the meter. If the audio signal chain is properly calibrated and has sufficient headroom, VU meters are very effective and non-fatiguing to the operator. VU meters are standard for on air consoles.

Peak Meters read and display the actual signal peaks. They typically respond nearly instantaneously to an increase in signal but fall slowly to enable the operator to follow the peak visually. Peak meters are often used for recording consoles to produce the best possible recordings.

d) TYPICAL PEAKS

The actual audio signal peaks above nominal level are a function of the type of music, the type of recording (tape or digital), the amount of compression, and other factors. Analog tape gradually saturates to produce maximum peaks about 14dB above 0VU. Digital recording can produce peaks 20dB above 0VU.

ANALOG & DIGITAL LEVELS

15.1 Digital Audio Levels

a) In analog audio systems, levels are specified by the nominal level (0VU, +4dBu) and a headroom specification that is determined by the audio equipment manufacturer. The headroom can be as little as 14dB and as high as 30dB depending on the application.

Typical headroom requirements are:

(1) Live Sound or Digital Recording	20dB
(2) Processed Mixdown	18dB
(3) Analog Tape Recording or Source	14dB

b) DIGITAL MAXIMUM LEVEL IS 0dBFS

Because all digital audio signals have a maximum output of all “1's”. it was decided to to define digital audio levels by their maximum output.

Maximum Digital Audio Output = 0dBFS (where FS means Full Scale)

c) WHAT IS THE NOMINAL LEVEL (0VU) IN A DIGITAL SYSTEM ?

Because 0dBFS is your reference when talking about digital levels, all other levels are negative with respect to 0dBFS. This is quite different from the way we discuss analog audio levels which are positive or negative relative to our reference point of 0VU.

If maximum output is 0dBFS, then the question in your specific audio system is what your nominal level (0VU) should be. Some digital recorders have these as calibration settings.

Typical nominal digital levels are:

(1) Live Sound or Digital Recording	-20dBFS	(highest headroom)
(2) Processed Mixdown	-18dBFS	(medium headroom)
(3) Analog Tape Recording or Source	-14dBFS	(lowest headroom)

The trade off in choosing among these settings is in noise floor versus possible distortion. The greater the headroom the higher the noise floor but the less likelihood of clipping the audio. In a broadcast facility, the higher headroom is usually the preferred choice so that work does not need to be repeated or a live show poorly recorded.

e) TAPE LEVEL VERSUS DIGITAL RECORDING LEVEL

Digital recordings are completely linear without any compression. Analog recordings to magnetic tape exhibit a natural compression due to tape saturation. This natural compression results in a tape recording sounding approximately 6dB louder than a digital recording at the same level.

ANALOG & DIGITAL LEVELS

15.2 Choosing & Calibrating Facility Levels

a) One of the most important decisions made in setting up a broadcast facility is the choice of a standard for levels throughout the facility. It then becomes critical to calibrate ALL equipment in the facility to match the chosen levels.

b) WHY IS CALIBRATION IMPORTANT?
Level variations from studio to studio is one of the greatest problems in broadcast facilities. As an example, if one studio records 0dBu as 0VU and another studio has 0VU calibrated at +4dBu, then an audio recording from the first studio will play back 4dB lower in the second studio. Also, audio recorded in the second studio will play back 4dB hotter in the first studio and may distort.

c) TYPICAL FACILITY ANALOG LEVELS
The most typical nominal analog facility level is +4dBu (1.228VRMS). This is because most recording equipment is calibrated for +4dBu as 0VU.

Some facilities will choose 0dBu (0.775VRMS) as the nominal facility level so that the console and source outputs have an additional 4dB of headroom before distortion. This choice reduces the Signal to Noise Ratio by 4dB. However, this may be inaudible if the studio noise floor is lower than the noise floor of your complete audio signal chain including transmitter.

d) INPUT AND OUTPUT LEVELS
Once a facility wide level has been selected (such as +4dBu), then the input and output levels of every piece of equipment in the facility should be set to receive that level and output that level. Refer to the manual for each product for specific information on proper calibration.

e) TYPICAL PROBLEMS
The typical calibration problem in an audio studio is created because of two or more adjustable gain stages in series. In one stage the signal is reduced while in the next it is raised. This continues two or more times until the audio signal is significantly degraded. This can occur in a signal chain with two or more products in series or within a single product (such as a console) that has two or more internal gain adjustments.

Most professional equipment can adjust its input gain to accommodate -10dBV consumer levels up to +8dBu professional equipment. It can also adjust its output level from -10dBV to +8dBu to accommodate various external equipment. This creates the potential problem for the signal level to be raised and lowered several times in a signal chain. This can introduce dramatic hum and noise.

As an example, calibrate a device to reduce the input level by 20dB and then raise the output level by 20dB. You can apply a +4dBu input signal to the device. It will reduce the input signal by 20dB to -16dBu and then add 20dB of gain at its output to raise the signal back from -16dBu to +4dBu. The audio level seems fine but hum and noise is added by the 20dB of internal gain. Put several pieces of equipment calibrated like this in a series and the audio performance will be entirely unacceptable. In the reverse case of gain followed by attenuation, the headroom will be reduced and significant distortion occur.

An example of a real world problem occurs with a console and a recorder. In this example the console output is calibrated at the correct station level. The console output feeds the input of a recorder. The recorder output is set for an attenuation of say -14dB (to take +4dBu down to -10dBV). The output of the recorder is connected to the input of the console. The console input channel is set to +14dB of gain to offset the attenuation in the recorder. In this situation, you can make an audio recording and it will play back correctly on the console. It will even play correctly in another studio. However, when played in the studio it was made in, the Signal to Noise ratio contributed by the console has been reduced by 14dB. All of the hum and noise in that channel has been raised by 14dB. This will usually be unacceptable.

Section 16

DECIBELS

16.0 Decibels, dBm, dBu, dBV, etc

a) Decibels are a very convenient unit of measure used to describe relative signal levels in audio systems.

DECIBELS (dB) = 20 x log(V1 / V2), where V1 and V2 are voltages

DECIBELS (dB) = 10 x log(P1 / P2), where P1 and P2 are power (watts)

b) WHY USE DECIBELS ?

Audio systems are made up of several audio devices in series. In working with them, we often need to calculate the signal at various points in the series. If we do not use decibels, then we must use multiplication to find the signal at each stage in the series. If we use decibels, we can simply ADD or SUBTRACT the numbers to find the signal level.

INPUT				OUTPUT
+4dBu	+40dB (X 100)	-20dB (/ 10)	+6dB (X 2)	+30dBu (X 20)

c) WHAT IS dBm ?

dBm is a decibel measurement where the P2 in the equation above is 1 milliwatt.

0dBm is a unit of measure for 1 milliwatt.

dBm is used to define a specific signal voltage (0.775VRMS) across a 600 ohm impedance.

The dBm measurement is traditionally used in telephone systems and has fallen out of common use in broadcast audio. It has been replaced by the dBu measurement which is a voltage only measurement and does not reference power or impedance.

d) WHAT IS dBu ?

dBu is a decibel measurement where the V2 in the equation above is 0.775VRMS (the voltage at 0dBm)

0dBu is a unit of measure for 0.775VRMS

+4dBu = 1.228VRMS (nominal level in professional audio equipment)

dBu has replaced dBm in audio measurements because modern audio interconnection systems are operated at a wide variety of impedances (not just 600 ohms).

e) WHAT IS dBV ?

dBV is used in consumer products for the same purpose as dBu in professional audio products

0dBV is a unit of measure for 1VRMS

-10dBV = 316 millivolts RMS (nominal level in consumer audio equipment)

f) WHAT IS dBr ?

The “r” in dBr means “RELATIVE.” In this type of measurement system, there will be a reference point and then other measurements are relative to that value. As an example, the reference point might be +4dBu at 1,000 cycles. The measurement at 20 cycles might be 0.1dBr and the measurement at 20,000 cycles might be -0.25dBr.

dBr is often used on digital voltmeters after you press a button to select a reference dB level.