AUDIO FUNDAMENTALS

Some basic concepts not to be forgotten by all audio professionals

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Things to retain in all audio manipulations

- Analogue amplitude units
- Dynamics in analogue and digital systems
- Analogue and digital interface formats
- Interfacing different systems

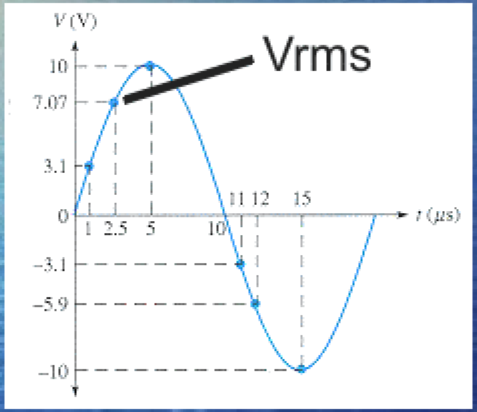
Analogue amplitude units

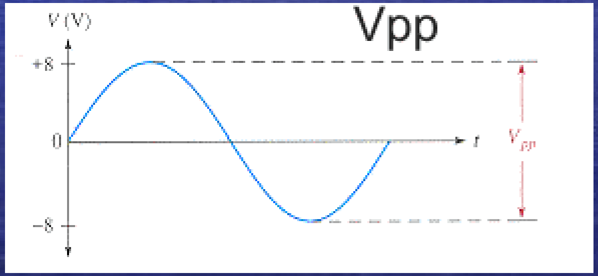
Volts

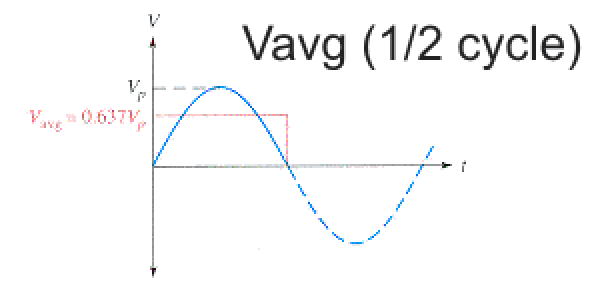
- By far the more commonly units used to express the magnitude of the signal present in audio interfaces.
- Widely used are sub-multiples mV and uV
- Vrms express the amplitude of a sinusoid which measures the same as the signal being measured
- For a sinusoid is the same as a DC voltage producing the same heat over a given resistor
- Vpp is defined as the total peak to peak amplitude of the signal
- For a sinewave, relation of VPP=2*sqrt(Vrms).
- For pink noise and music, this relationship considerably increases.

Analogue amplitude units

Vrms, Vpp, Vavg







Relative amplitude units

dB, dBm

- dB is a relative unit, so its use is only possible against a known reference. It is not by all means an absolute unit. In voltage or intensity terms
- $dB=20*\log_{10}(Vmeasured/Vref)$

or in power terms

- $dB=10*\log_{10}(Pmeasured/Pref)$

Widely used in valve equipment times was the dBm. Definition of 0dBm is the voltage producing 1mW over a known impedance, which used to be 600Ω. This explains that as P=V2/R and consequently V=√P*R, 1mW over 600Ω is 0.775V, or 775mV. This level is still widely used today, but more referred to as dBu, because reference to a 600Ω impedance has become obsolete.

Relative amplitude units

dbu, dBV, dBr, dBFS

- dBu measures the signal against a reference of 775mV. So, -6dBu is 388mV, and +6dBu is 1.55V.
- dBV measures the signal against a reference of 1V. So, -6dBV is 0.5V, and +6dBV is 2V (a widely used level in what CD players are concerned, for example)
- 0dBV is +2.2dBu
- 0dBFS is the maximum available level of a digital system, when all available bits have been used.
- When the reference is arbitrary, we use the term dBr re=Vref
- Never forget that double voltage or current is +6dB, and double power +3dB

Noise, saturation and dynamic range

Analogue system considerations

- If the reference level of an audio system is 0dBu and the noise floor is at -80dBu, signal-to-noise ratio for a signal of 0dBu is 80dB.
- If the same system is able to allow without saturation a level of +22dBu, total dynamic range is 80+22=102dB. This is the total possible excursion of the signal from the noise level to the distortion one.
- If the signal is too weak we will hear the noise, so we will have to increase it, but if too strong, it will be distorted.
- To avoid both extremes, compression and limiting is often used. As compression and limiting may exhibit side effects as pumping, it must be used with care.

Noise, saturation and dynamic range

Digital system considerations

- While saturation in an analogue system can be tolerated to a certain degree, in a digital system it must be avoided at all cost, because of its extremely annoying character (bit-reversal, for example).
- Noise in a digital system has sometimes non-gaussian contents, like embedded tones in early sigma-delta converters.
- The two previous considerations dictate that we must get away from both extremes.
- Processing, particularly in non-floating point DSP systems may impair audio resolution by coefficient scaling, degrading the noise floor and introducing artifacts as aliasing and pre-echoes.

Noise, saturation and dynamic range

Digital versus analogue

- A system with a -98dBu noise floor and a +22dBu saturation point is a very good one in analogue terms (120dB dynamic range). It is relatively easy to obtain that performance using current technology.
- The basic dynamic range of a digital system, taking out the uncertainty bit will be $20*\log(2^{\text{bit-width-1}})$.
- To duplicate an analogue system such as the one described before we will need at least a 24-bit system.
- Dither, which is in fact an extraneous signal introduced to spread the noise spectrum, can be used to improve the performance at low levels .

Noise, saturation and dynamic range

Digital system dynamic range

As each bit relates to the previous one by a power of 2, 6dB is lost by taking one bit out of the total, either on the saturation or on the noise side

Bit-width	Dynamic range(dB)
16	91
18	103
20	115
24	139

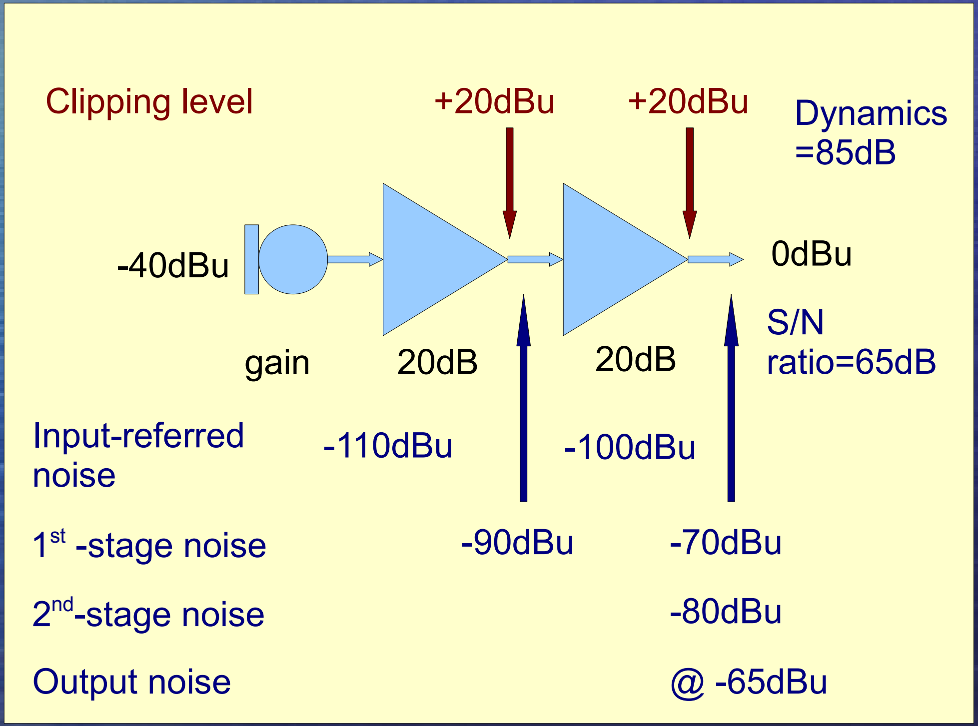
.

Common levels in audio systems

- Line level inputs may range from -20dBu to +20dBu, +4dBu being the most common.
- Microphone levels will typically range from -60dBu to -20dBu
- Note that the reference level of a digital system is not 0dBFS, this to avoid that a recording signal hits the maximum bit-width and saturates, exhibiting gross distortion.
- If we choose to have a 18dB overload margin, setting the digital system level at -18dBFS we are loosing 3 bits there, and one more at the noise level, so calculate 4 bits less than the advertised bit-width
- If you have a limited bit-with system, compress and raise the level to escape from noise, using a good analogue/digital system with a resolution greater than the advertised bit-with of the digital system which is used as the final media.

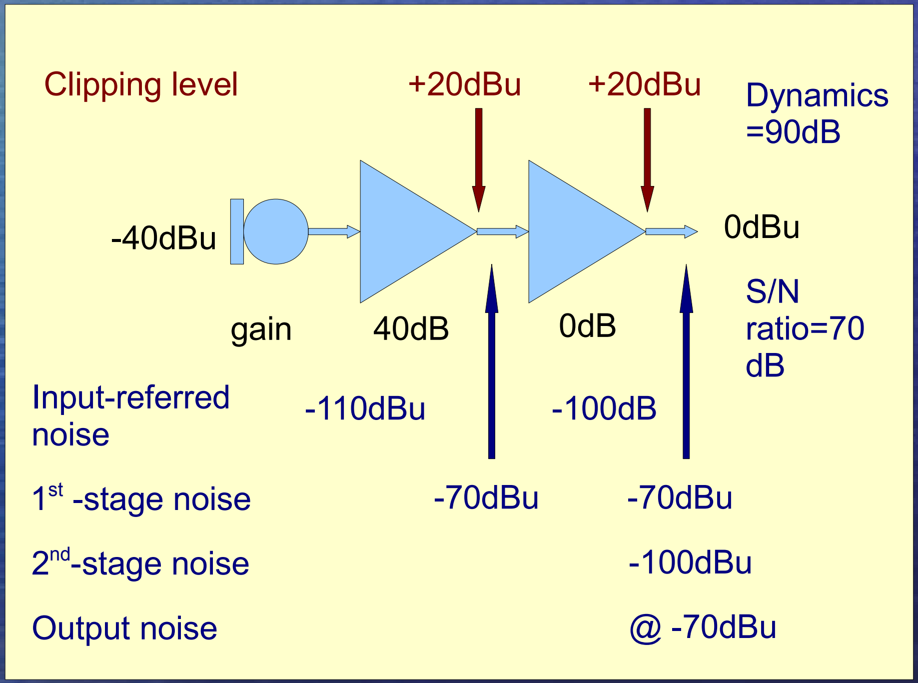
A practical example

A microphone preamplifier followed by an output stage, splitting the gain between stages, a bad practice...



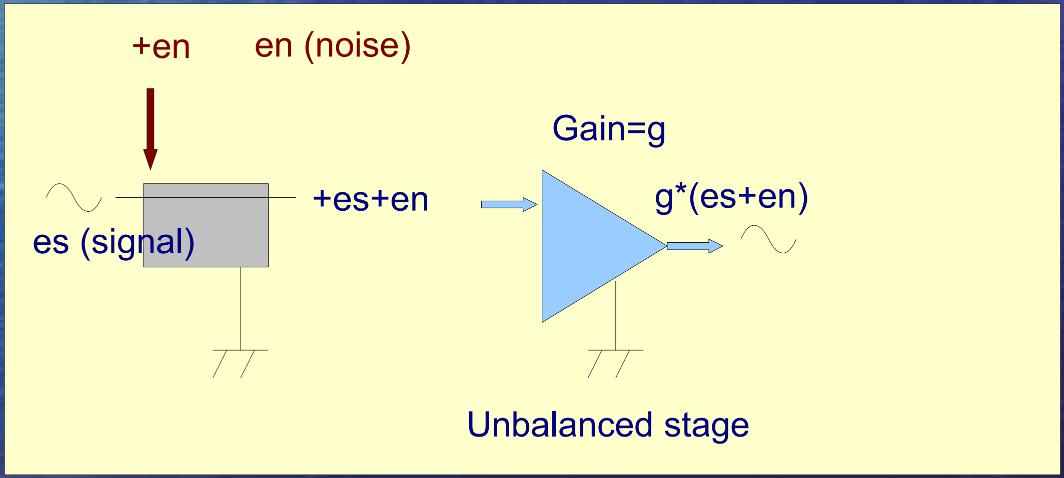
A practical example

A microphone preamplifier followed by an output stage, concentrating the gain where dynamics is greater, a good practice...



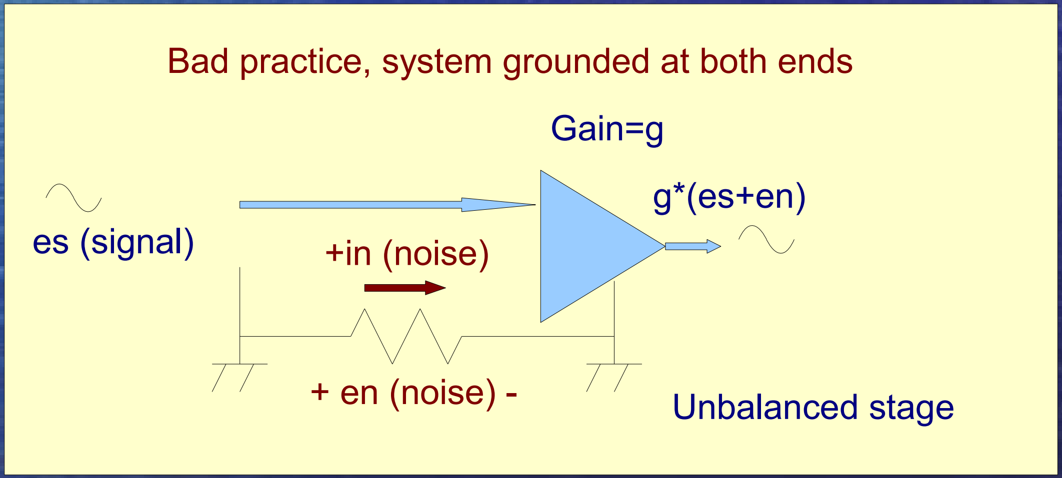
Unbalanced analogue Interfaces

- Unbalanced analogue interfaces use one signal wire and ground.
- If noise is picked up by the signal wire, it will appear at the output, amplified by the circuit gain.



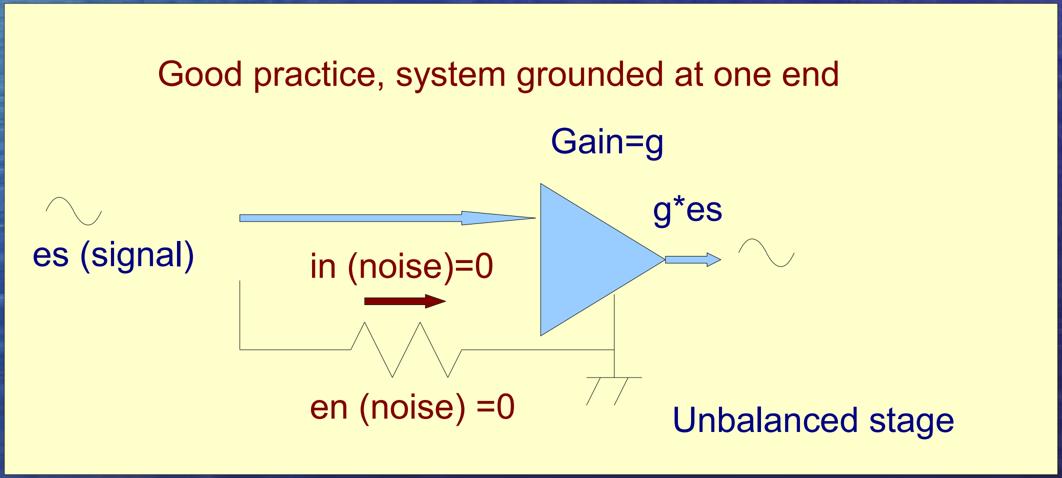
Unbalanced analogue Interfaces

- Another potential problem appears when a system is grounded in several different places
- Noise currents between grounds will circulate through finite ground resistance and be amplified as noise.



Unbalanced analogue Interfaces

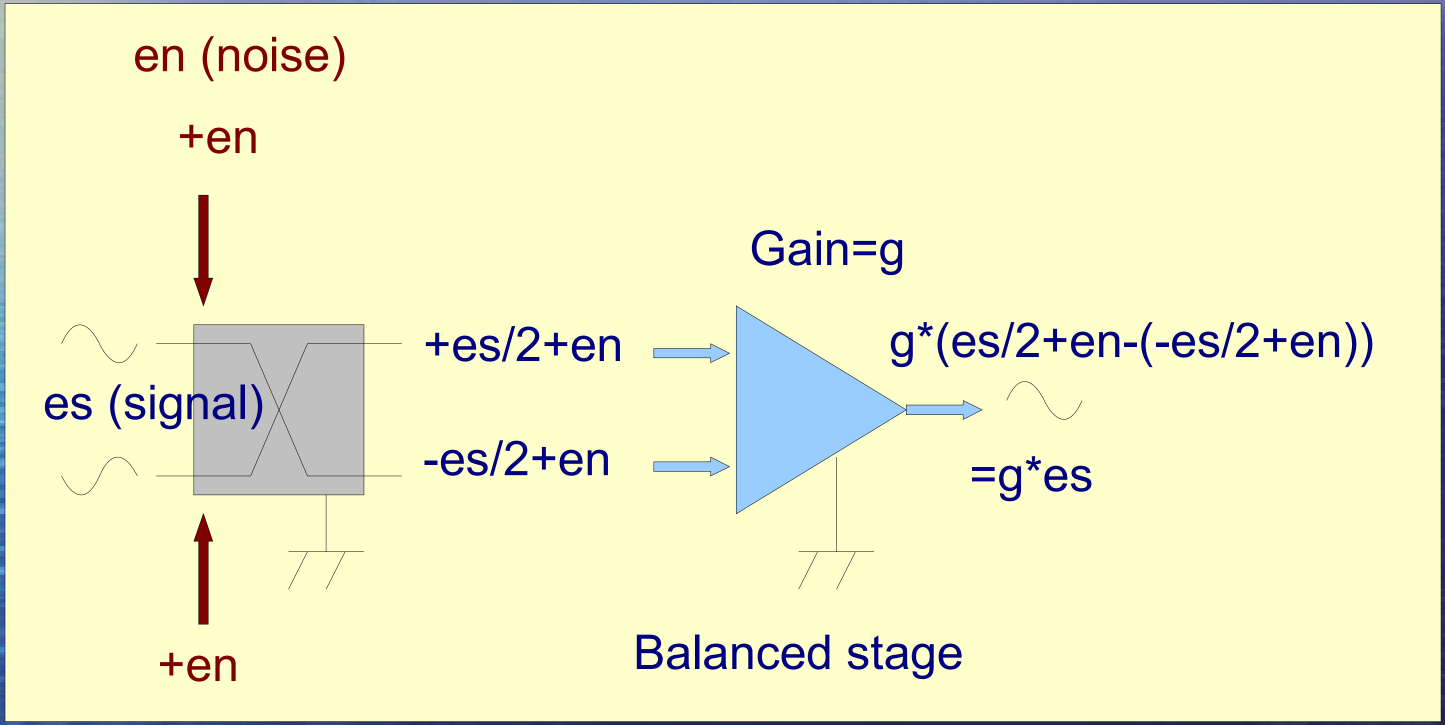
- If a system is grounded just at one of the ends, ground problems are avoided
- In a system composed of several stages, just one of those shall be grounded.



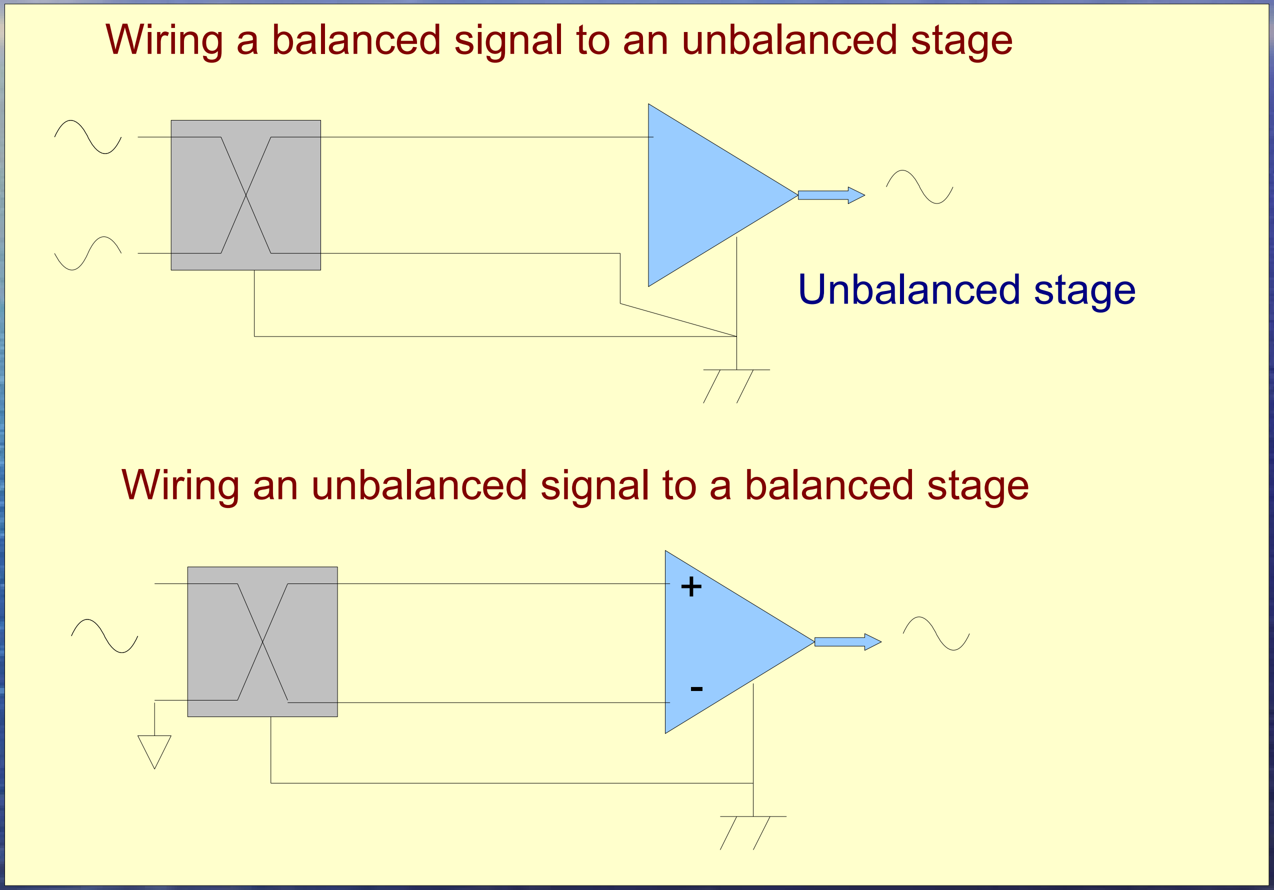
Balanced analogue Interfaces

- Balanced analogue interfaces use two out-of phase signal wires.
- If noise is equally picked up by the two signal wires, it will theoretically be ruled out.
- That does not happen, common-mode signals are always amplified, because cables and amplifiers are not perfect. The relation of the differential (useful) signal to the common-mode (noise) signal is called the Common Mode Rejection Ratio (CMRR) and the higher the better.
- CMRR tends to vary with frequency, degrading at extremes. Because of ground problems, it is important to ensure that at low frequencies CMRR is high (>60dB)

The balanced interface



Wiring from bal/unb/bal

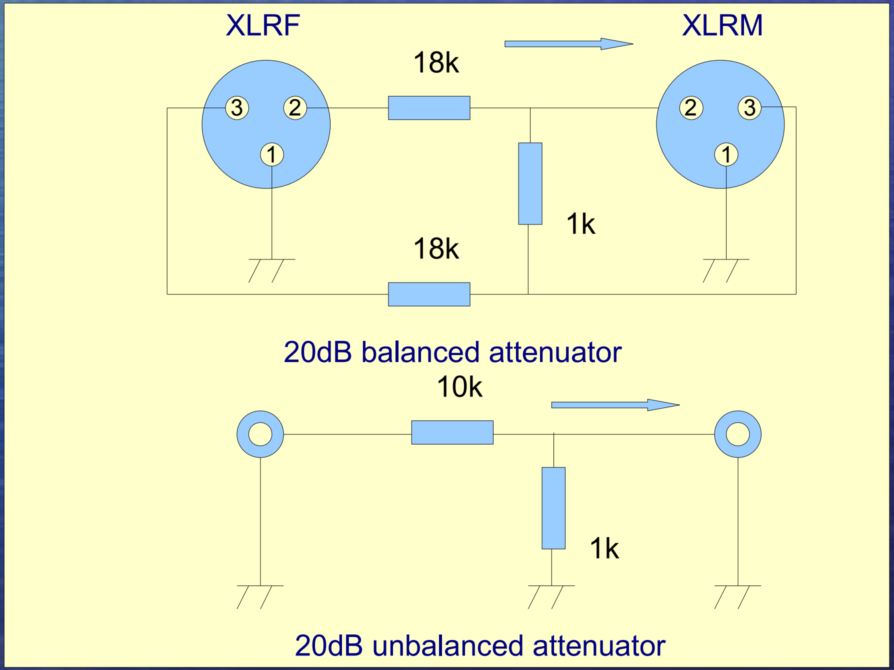


A slightly annoying problem

- Suppose you have an amplifier to which you want to connect a -20dBu signal. That amplifier has two switchable fixed-gain inputs:
  - a microphone input (-50dBu, overload at -30dBu)
  - A line input (-10dBu, overload at +20dBu)
- Amplifier input impedance is 100kOhm
- If you feed the signal to the microphone input, it will overload it, and the only thing you will be able to extract even with the volume down is distortion. If fed to the line input, our signal won't drive the amplifier properly to full power...

The answer, a 20dB attenuator

- That would reduce the signal to -40dBu, which is well within range of the microphone input, and still allow for a 10dB headroom before saturation.



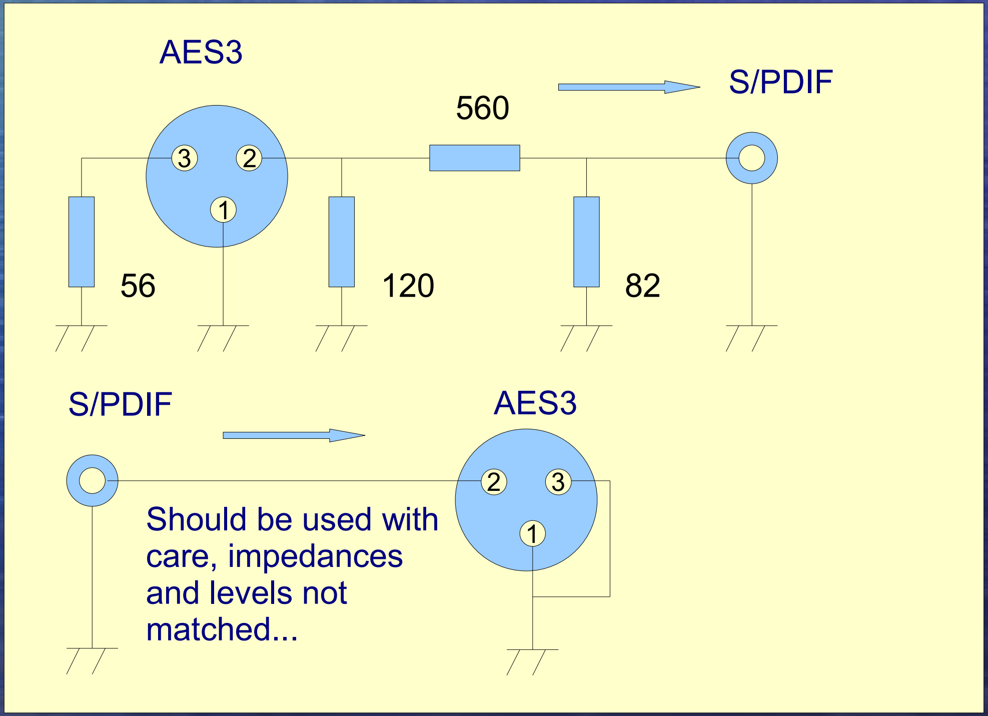
Digital Interface formats

- AES3 and S/PDIF are the most common digital interfaces.
- AES3 is electrically a balanced interface and S/PDIF is an unbalanced interface. All the preceding remarks about analogue interfacing are applicable to digital interfaces. Noise can impair transmission of digital signals and cause jitter.
- S/PDIF signals can be wired straight into an AES3, reverse needs an attenuator.

	AES3	S/PDIF
Interface	Balanced	Unbalanced
Connector	XLR-3	RCA
Impedance	110 ohms	75 ohms
Output Level	2-7 Vp-p	0.5 Vp-p
Max Output	7 Vp-p	0.6 Vp-p
Max Current	64 mA	8 mA
Min Input	0.2 V	0.2 V
Cable	STP	Coax
Max Distance	100 m	10 m

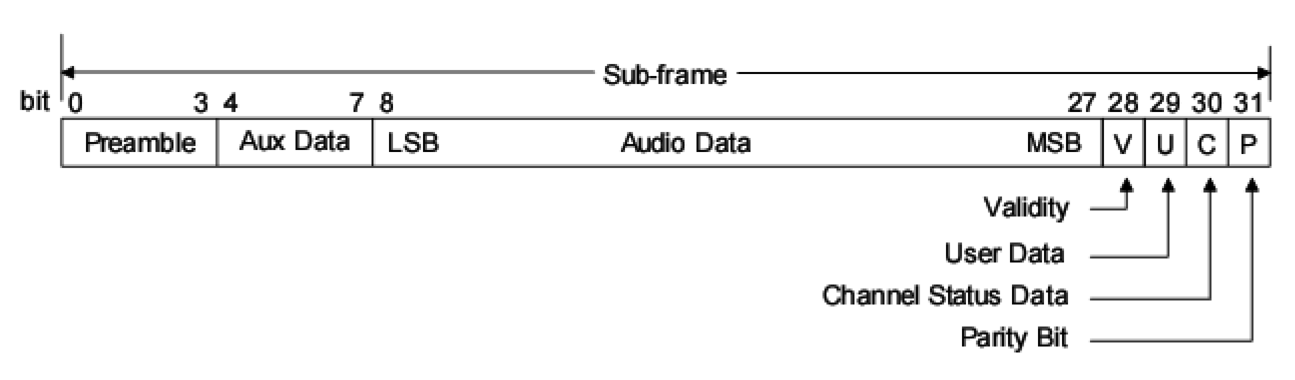
Digital format conversion

- A transformer adapter is always the best match. But if you don't have one around...



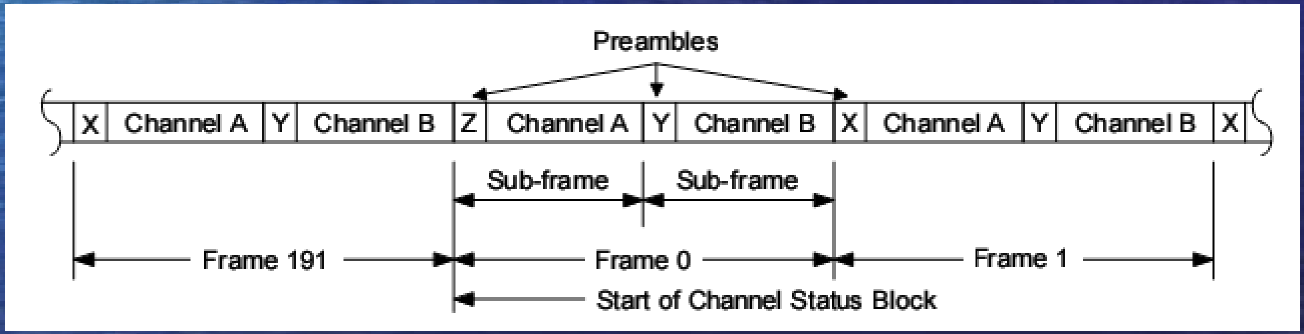
What's inside AES3 frames

- Data is transmitted bi-phase (to pass though isolation transformers)
- A preamble (unique transmission) is used to identify frame start
- Then some ancillary data is transmitted (4 bits), and then audio and some special status bits

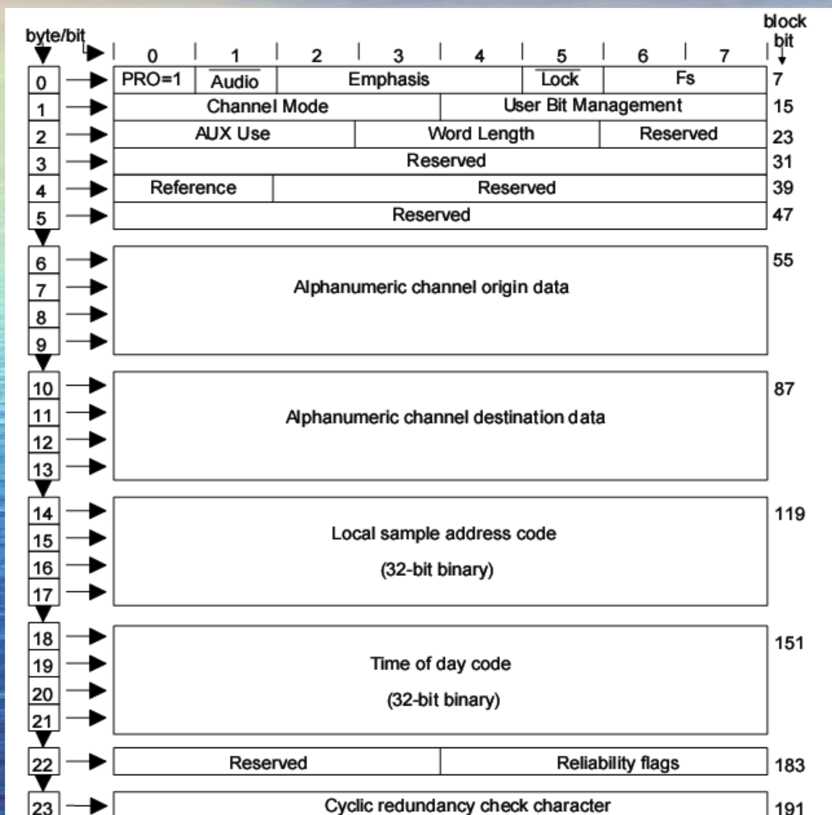


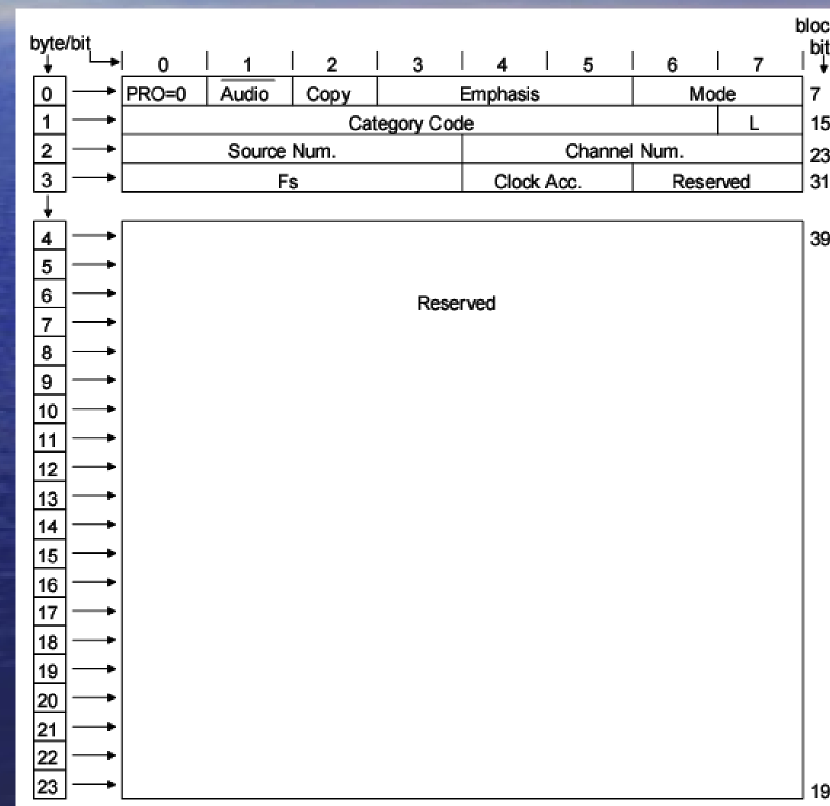
What's inside AES3 frames

- Left and right frames are transmitted sequentially
- Different preambles are used for the decoder to locate the data
- Ancillary data which is transmitted 4 bits at a time will be recovered each 192 frames



Professional versus consumer block





What's inside the professional block

BYTE 0		
bit	0	**PRO = 0 (consumer)**
	0	Consumer use of channel status block
	1	Professional use of channel status block
bit	1	**Audio**
	0	Digital Audio
	1	Non-Audio
bits	2	**Copy / Copyright**
	0	Copy inhibited / copyright asserted
	1	Copy permitted / copyright not asserted
bits	3 4 5	**Pre-emphasis - if bit 1 is 0 (dig. audio)**
	0 0 0	None - 2 channel audio
	1 0 0	50/15 µs - 2 channel audio
	0 1 0	Reserved - 2 channel audio
	1 1 0	Reserved - 2 channel audio
	X X 1	Reserved - 4 channel audio
bits	3 4 5	**if bit 1 is 1 (non-audio)**
	0 0 0	Digital data
	X X X	All other states of bits 3-5 are reserved
bits	6 6	**Mode**
	0 0	Mode 0 (defines bytes 1-3)
	X X	All other states of bits 6-7 are reserved

BYTE 1 - Category Code 001		
bits	3 4 5 6	**Broadcast reception of digital audio**
*	0 0 0 0	Japan
*	0 0 1 1	United States
*	1 0 0 0	Europe
*	0 0 0 1	Electronic software delivery
	X X X X	All other states are reserved

BYTE 1 - Category Code 100		
bits	3 4 5 6	**Laser Optical**
	0 0 0 0	CD - compatible with IEC-908
*	1 0 0 0	CD - not comp. with IEC-908 (magneto-optical)
	X X X X	All other states are reserved

BYTE 1			
bits	0 1 2 3	4 5 6	**Category Code**
	0 0 0 0	0 0 0	General
*		0 0 1	Experimental
		X X X	Reserved
*	0 0 0 1	X X X	Solid state memory
*	0 0 1 X	X X X	Broadcast recep. of digital audio
	0 1 0 X	X X X	Digital/digital converters
*	0 1 1 0	0 X X	A/D converters w/o copyright
*		1 X X	A/D converters w/ copyright (using Copy and L bits)
*	0 1 1 1	X X X	Broadcast recep. of digital audio
	1 0 0 X	X X X	Laser-optical
*	1 0 1 X	X X X	Musical instruments, mics, etc.
	1 1 0 X	X X X	Magnetic tape or disk
	1 1 1 X	X X X	Reserved
bit	7		**L: Generation Status.**
			Only category codes:001XXXX, 0111XXX,100XXXX
*	0		Original/Commercially pre-recorded data
*	1		No indication or 1st generation or higher
			All other category codes
*	0		No indication or 1st generation or higher
*	1		Original/Commercially pre-recorded data

The subgroups under the category code groups listed above are described in tables below. Those not listed are reserved.

The Copy and L bits form a copy protection scheme for original works. Further explanations can be found in the proposed amendment (TC84) to IEC-958.

BYTE 1 - Category Code 010		
bits	3 4 5 6	**Digital/digital conv. & signal processing**
	0 0 0 0	PCM encoder/decoder
*	0 0 1 0	Digital sound sampler
*	0 1 0 0	Digital signal mixer
*	1 1 0 0	Sample-rate converter
	X X X X	All other states are reserved

BYTE 2			
bits	0 1 2	**AUX: Use of auxiliary sample bits**	
	0 0 0	Not defined. Maximum audio word length is 20 bits	
	0 0 1	Used for main audio. Maximum audio word length is 24 bits	
	0 1 0	Single coordination signal. Max. audio word length is 20 bits	
	0 1 1	User defined application	
	X X X	All other states of bits 4-7 are reserved.	
bits	3 4 5	**Source word length Max. audio based on bits 0-2 above**	
		Max audio 24 bits	Max audio 20 bits
	0 0 0	Not Indicated	Not Indicated (default)
	0 0 1	23 bits	19 bits
	0 1 0	22 bits	18 bits
	0 1 1	20 bits	16 bits
	1 0 1	24 bits	20 bits
	X X X	All other states of bits 3-5 are reserved	
bits	6 7		
	X X	Reserved	

BYTE 3		
bits	0-7	**Vectored target byte**
	XXXXXX	Reserved

BYTE 4		
bits	0 1	**Digital audio reference signal per AES11-1990**
	0 0	Not reference signal (default)
	0 1	Grade 1 reference signal
	1 0	Grade 2 reference signal
	1 1	Reserved
bits	2 7	
	XXXXXX	Reserved

BYTE 5		
bits	0-7	
	XXXXXX	Reserved

BYTE 6-9
Alphanumeric channel origin data
7-bit ISO 646 (ASCII) data with odd parity bit. First character in message is byte 6. LSB's are transmitted first.

BYTE 10-13
Alphanumeric channel destination data
7-bit ISO 646 (ASCII) data with odd parity bit. First character in message is byte 10. LSBs are transmitted first.

BYTE 14-17
Local sample address code (32-bit binary)
Value is of first sample of current block. LSBs are transmitted first.

BYTE 18-21
Time-of-day sample address code (32-bit binary)
Value is of first sample of current block. LSBs are transmitted first.

BYTE 22		
bits	0 1 2 3	
	X X X	Reserved
bit	4	**Channel status bytes 0 to 5**
	0	Reliable
	1	Unreliable
bit	5	**Channel status bytes 6 to 13**
	0	Reliable
	1	Unreliable
bit	6	**Channel status bytes 14-17**
	0	Reliable
	1	Unreliable
bit	7	**Channel status bytes 18 to 21**
	0	Reliable
	1	Unreliable

BYTE 23
CRCC: Cyclic redundancy check character
CRCC for channel status data block that uses bytes 0 to22 inclusive. Generating polynomial is $G(x) = X^8 + X^4 + X^3 + X^2 + 1$ with an initial state of all ones

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*	1 1 0 0	Sample-rate converter
	X X X X	All other states are reserved

BYTE 1 - Category Code 101		
bits	3 4 5 6	Musical Instruments, mics, etc.
*	0 0 0 0	Synthesizer
*	1 0 0 0	Microphone
	X X X X	All other states are reserved

BYTE 1 - Category Code 110		
bits	3 4 5 6	Magnetic tape or disk
	0 0 0 0	DAT
*	1 0 0 0	Digital audio sound VCR
	X X X X	All other states are reserved

BYTE 2		
bits	0 1 2 3	Source Number
	0 0 0 0	Unspecified
	1 0 0 0	1
	0 1 0 0	2
	1 1 0 0	3
	0 0 1 0	4 to
	0 1 1 1	14 (binary - 0 is LSB, 3 is MSB)
	1 1 1 1	15
bit	4 5 6 7	Channel Number
	0 0 0 0	Unspecified
	1 0 0 0	A (Left in 2 channel format)
	0 1 0 0	B (Right in 2 channel format)
	1 1 0 0	C to
	0 1 1 1	N (binary - 4 is LSB, 7 is MSB)
	1 1 1 1	O

BYTE 3		
bits	0 1 2 3	Fs: Sample Frequency
	0 0 0 0	44.1 kHz
	0 1 0 0	48 kHzr
	1 1 0 0	32 kHz
	1 1 0 0	Sample-rate converter
	X X X X	All other states are reserved
bits	4 5	Clock Accuracy
	0 0	Level II, ±1000 ppm (default)
	0 1	Level III, variable pitch
	1 0	Level I, ±50 ppm - high accuracy
	1 1	Reserved
bits	6 7	
	X X	Reserved

BYTE 4 - 23
Reserved

Thanks for your patience...