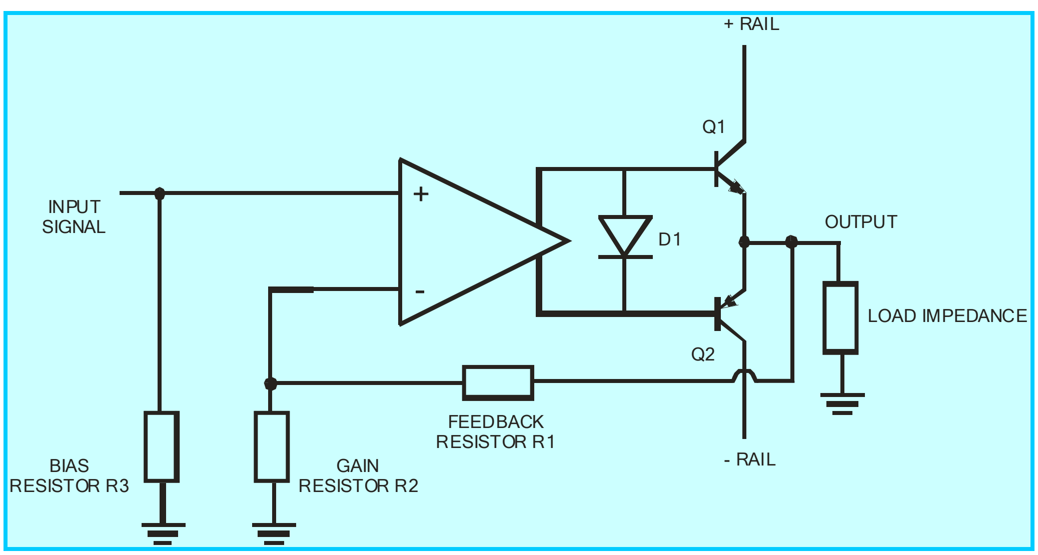
Amplifier clipping

Amplifier clipping – What is it? How is it caused and how can we prevent it? What does it do? We shall deal with each of these topics one at a time.

What is it: Almost everyone assumes that amplifier clipping is the sole domain of power amplifiers? This is not true. A preamplifier is just as prone to clipping as power amplifiers. If the level of signal is high enough to cause the preamplifier to clip, the power amplifier being a faithful servant will just amplify the clipped signal it receives. For the purposes of this discussion we will assume that our amplifier/preamplifier models use a bi-polar power supply (as almost all audio electronics does today) and therefore the signal swings from a level of zero to either the positive or negative supply rails ("rail" is a commonly used term which we use to describe a power supply output). We shall also assume that the electronic building blocks are in the form of operational amplifiers WITH negative feedback. Op-amps as they are called, are TWO input ONE output building blocks. Input is to either positive or negative ports and feedback is taken from the output and returned to the (-) input. I shall show the op-amps as in the diagram below even though it is not the standard schematic symbol for an op-amp.

Note: A power amplifier (that is one which can drive a loudspeaker) is nothing less than just a high current preamplifier. Preamplifiers can and do run off very high rail (there we go again with the term "rail") voltages – they just do not have the capability to source lots of current.



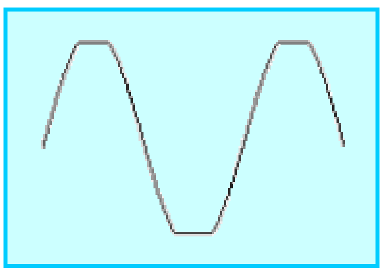
The gain of this amplifier is defined as the ratio of R1+R2/R2. So if R1=10K ohm and R2 = 1K ohm the gain is 10,000+1,000/1000 = 11 times. So whatever signal is applied to the input, it shall be amplified 11 times PROVIDED the output signal does not exceed either rail voltage. Let's assume the + and – rails are each 30v. This means that the output rail can move towards either the +30 or the -30 volt rails depending on whether Q1 or Q2 is turned on. They CANNOT both be conducting at the same time. The diode D1 is shown as the source of bias which sets up the idling current for the class A-B output stage. I have just shown it to represent the fact that this building block of ours is a class A-B output stage.

Note: We can use the peak value of the input signal versus the peak value of the output signal but this does not equate to the real world so I shall use the RMS volts in versus RMS volts out. 1 volt peak = 0.7071 volt RMS. We shall assume perfect transistors for Q1 and Q2 so they have no losses. This means that with the 30 volt rails, the maximum RMS output voltage is 30 x 0.7071 = 21.2 volts.

Now the above schematic could represent a preamplifier stage or a power amplifier. The difference is simply the value of the LOAD IMPEDANCE we place at the output. If this was a preamplifier the load impedance would be in the order of thousands of ohms and if it was a power amplifier it would be say 4 ohms.

Note: We are also assuming that the rails have infinite current capability (the 30 volts remains constant) and that the transistors Q1 and Q2 can pass any current we require.

If we keep the gain at 11 times this means that the maximum input voltage allowed in order to prevent clipping is 21.2/11 = 1.93 volts. Applying more than 1.93 volts to the input will cause the transistors Q1 and Q2 to “bump” their heads at their respective rails. This is known as clipping. The waveform at the output of our amplifier looks like the diagram as shown below



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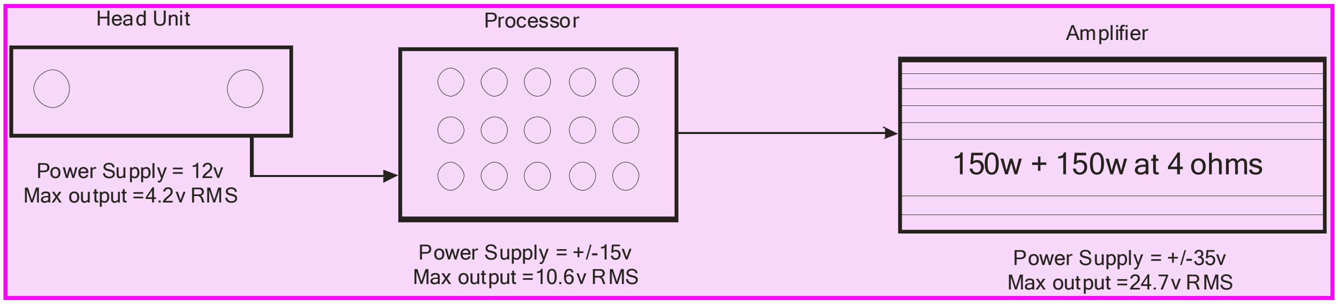
As we see the top and bottom of the sine wave has been flattened. When the waveform rises (or falls) to it’s respective rail and it reaches a value of 21.2v RMS (30v peak) it cannot increase/decrease any further.

If we change the gain of the amplifier to say 20 times, (change R1 or R2 in the abive circuit) the only difference is that we shall require less input voltage to drive the amplifier to a given output voltage. Since we know that the maximum output before clipping is 21.2 volts, dividing this by 20 yields a value of 1.06 volts at the input. Conclusion, clipping is not affected by the initial design of the gain structure (since we can apply higher input voltages if the gain is low and visa versa) but only by the value of the power supply voltages.

How is clipping caused.

Simple by allowing the output stage of a preamplifier or power amplifier’s output voltage to reach the value of the power supply voltage. A typical mobile audio system consists of a head unit, an equalizer and then an amplifier. It looks like the diagram below.
We shall begin with the head unit. Working from a 12v nominal power source (The battery) it is capable of delivering a theoretical output of 12/2 = 6v peak now x0.7071 =4.24v RMS



Typically this output voltage is limited to slightly less than 4 volts RMS. So the first stage of protection is to ensure that the output of the head does not exceed this nominal 4 volts. How do we do this? The only sure way is to use your favourite CD which has the loudest passages and connect an oscilloscope to the RCA outputs and confirm that with the volume control set at maximum, there is no clipping of the complex music waveform. By setting the timebase control on the 'scope it is very easy to see clipping. We shall assume for the moment that the head unit manufacturer has designed the product correctly and that the highest modulated CD will not allow any form of clipping at the outputs.

Now for the tricky part. We have a head whose output varies from 0 to 4 volts depending on the volume setting. At 0 volts output we will not induce clipping anywhere!!!

The processor which works off a typical split rail of +/-15 volts can output in practice 9v RMS. Normally these units have some sort of gain controls and for this example let us assume it only has an input level control (See also the links on Level Control and Level Matching). The max. of 4v being the worst case from the head now enters the processor. Hopefully the input level control is in the correct place in the processor's signal path and we shall assume this to be true. This level control obviously changes the **gain structure** (how many times the processor will amplify the in-out signal level). We shall assume that it can change the gain from -10dB to +10dB.

Note: Decibels (dB) is a logarithmic way of denoting a ratio. 10dB in voltage terms is calculated from this formula dB = 20 x log to the base 10 of the ratio. So +10dB in voltage terms is 3.16 times. -10dB is 0.316 times.

Back to our processor. All crossover controls and equalizer settings are flat. So the equalizer has no effect now and the crossover settings are such that we are measuring voltages within a particular passband. (Let's assume it is a 2 way crossover set at 100Hz and we are testing a signal at 1KHz which is far away from the 100Hz at nearly 4 octaves). Setting the gain at 0dB we would get 4 volts out of the processor which is well below its clip point of 9v. If we set the processor's level control to any position less than 0dB we are still in a non clipped mode. Moving this level control to the +dB side we start moving towards the "red" zone. Remember that we have 4v from the head. The processor can output 9v, this means a **MAXIMUM** gain structure of 9/4 = 2.25 times! (2.25x = 7dB).
So we can only set our processor's level control to +7dB before the onset of clipping. What we have in effect done is put a safety net up by setting the processor's gain to +7dB we guarantee ourselves that NO clipping can occur no matter where the level control of the head is set.

Note: What we have to know is the absolute max output of the head at max volume setting WITHOUT clipping. This is our start point and if it is incorrect the rest of the exercise is pointless!

So now we have a processor which can output up to 9v with NO clipping. Our 150w/ch amplifier is the next victim. Let us assume (yes we must assume things for examples) that our amplifier has a level control whose range is 200mV to 9v for rated output of 24.7v.

Note: Please do not be confused with my references to an amplifier's output voltage and not its power. The amplifier is just a **great big old preamplifier** which can source lots of current but at the end of the day it still puts out VOLTS and sinks AMPS. It is easier when referring to gain structures and voltage gains to use volts at each end of a piece of gear. There is NO reason in the world why we cannot specify a preamplifier's output as power. It may be very small but it still is power. You will notice that Zed Audio specifies the output of the amplifier in volts, amps and then watts. The volts/amps is a more accurate way of describing what the amplifier is capable of.

If we set our amplifier's level control to 9v (We assume the amplifier puts out a wee bit more than 24.7v before clipping) then we still have a system which does not clip. (Do not forget that we set the processor level to +7dB to have a 9v output)

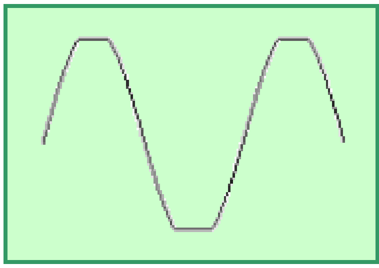
So we now can repeat the same thing with the amplifier. Setting its level control to say 2 volts will 100% guarantee that we can clip the amplifier with ease. How do we prevent this from happening. First do not power up the system – no only joking! First make sure that the level control on the processor is then set back so that with 4v in from the head it can then only output 2 volts. Therefore we must set the level control of the processor to 2/4=0.5 times = -6dB.

Note: When boosting ANY equalizer control by "n"dB this is equal to lifting the gain by "n"dB. Even though the equalizer boosts the level in a relatively narrow band of frequencies the amplifier does NOT differentiate. So this "n"dB must be taken into account and the level should be reduced by "n"dB at either the processor or amplifier. Processors with both input and output level controls are far better at level matching than those with only one level control.

For more on this subject see the link on Level Matching.

What does it do?.

Clipping is the arch enemy of speakers, especially higher frequency drivers. It is probably the biggest cause of speaker failure. Looking at the diagram below which shows a clipped sinewave we see from the time axis that the waveform remains at a high amplitude (either positive or negative) for a period of time which is longer than the time it spends when the sinewave is not clipped.



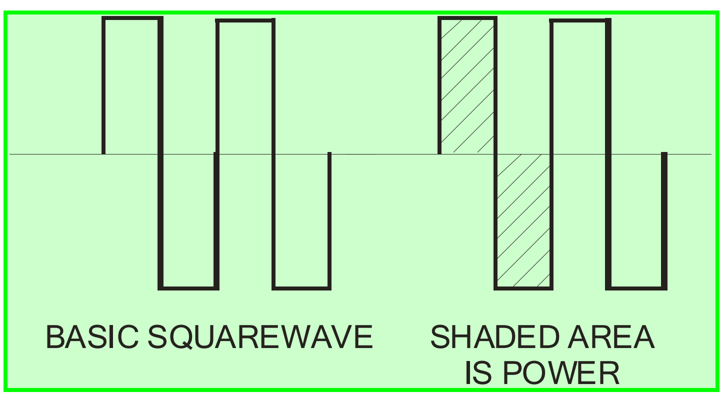
Amplitude

Time

The result of the speaker cone “spending” too much time at one end of its travel will cause voice coil overheating, deformity of the cone/spider assembly. Another effect of amplifier clipping is that harmonics are generated from the fundamental. Assume a 100Hz wave is being clipped. Harmonics at 200Hz, 300Hz, 400Hz, etc are generated. As the harmonic number increases, its amplitude decreases. The amplitude of these higher frequency harmonics is determined by how hard the amplifier clips at the fundamental frequency.

Because high frequency drivers are fragile as compared to high power low frequency and midrange drivers, they are more susceptible to damage. These high frequency harmonics do not generally damage low frequency drivers but this is not always 100% true.

Let us use a 200 watt amplifier as our example and let it be clipping at say +6dB worth of overdrive. +6dB of overdrive in power terms is calculated from the formula [dB=10 x log to the base 10 x power ratio]. Putting the numbers in the formula yields an answer of 4 times power. So the 200 watt amplifier will “attempt” to put out 800 watts. When an amplifier is hard clipped it puts out essentially a square wave which looks like this:



The area under the squarewave represents power and if one compares this with a sinewave at the same frequency, then it is obvious that the area under a sinewave is much less than the square wave.

Music is not constant in its peak amplitude. The ratio of average power to peak power is in the order of 10-20dB. (10dB = 10 times power and 20dB = 100 times power). I would imagine that modern rock and roll/rap music the value is closer to 10dB. This means that with typical music the average power when using a 200w/ch amplifier is in the order of 20 watts per channel with the peaks rising to 200 watts. Anything higher than the 20 watt average will most certainly push the amplifier into clipping. With this scenario the tweeter in a typical bi-amplified system or one with passive crossovers will receive about 10-15% of the power. So the tweeter’s power is about 20-35 watts with our 200 watt amplifier. This is a lot of power for any single tweeter. But let us assume it is OK with this.

When the amplifier clips the energy into the tweeter is many times greater than with unclipped signals. (Of course the amount depends on the degree of clipping but it has been found that people will listen up to 10dB of clipping). When this happens the compressed wave (now very close to a square wave) is absorbed by the tweeter (and do not forget about all the harmonics) and at this stage the tweeter goes to “the pie in the sky”.

Low frequency drivers are more tolerant of clipping simply because of their more robust construction. I have however seen many a woofer damaged through been overloaded on a continuous basis.

The above discussions have assumed that the waveform is symmetrical about the zero line. Unfortunately music is not like this. The positive half of the wave may not be the same as the negative half. As an example let us assume that this is so and that the positive part of the wave at time zero is larger in amplitude than the negative half. When the amplifier clips, the area under the positive half is more than the negative half and because square waves are being generated by the amplifier the DC component on the speaker rail will not be zero – as it should be.

Remember one fact. DC is a constant voltage. 10 volts positive DC (ref zero) is just that. If our amplifier was flat to DC and we put in a DC signal the amplifier would simply do it’s job – amplify and the output at the speaker rail would be a larger replica of the input. AC on the other hand is just varying DC. A sinewave begins at 0 volts. It rises at a particular rate (determined by the frequency) to its peak value and then declines to zero and repeats the same thing below the zero ref line. BUT at any given time during the single cycle of the sinewave it has an absolute value. The average is zero. A square wave (clipping!!!!) is similar but not the same. The square wave starts at zero, rises very rapidly to it’s peak value, stays there for a time (determined by the frequency) and then returns to zero and the other half of the cycle is below the zero line. The average of course is zero ONLY if the positive half of the square wave is equal to the negative half.

With music and clipped amplifier the average is not zero and in our example above the speaker rail will tend to move positive DC for the period of that non symmetrical clipped wave. DC on a speaker for a sustained period of time (Constant amplifier clipping) will sustain damage.