Natural Language is the New User Interface

Colin Davies, Applications Manager Europe, Diodes Inc.

The 'perfect storm' of high bandwidth internet connectivity, deep learning algorithms and high quality audio is enabling a new paradigm in user interfaces: the spoken word. The ability to interact with household appliances and cloud-based services through natural language is truly a sea-change in the way people and machines interact.

Digital assistants may seem like the pinnacle of technology today, but really they are just the beginning of the next generation of user interface. Speech recognition, as a technology, has been around for many years; there are numerous software packages designed to turn speech into text, for example. The difference is how the software interprets the text. To date, it has simply been relayed to a screen, but more recently the machines seem to understand the intent, as well as the words. This level of artificial intelligence is going to become pervasive over the next few years, which means it will be easier to access and more attractive to integrate into everyday items.

The challenge for manufacturers will be how to add a high level of audio quality to ever-smaller electronic devices. Wearable technology is a trend developing in parallel with AI and natural language interfaces, so combining the two will demand a higher level of integration in order to preserve fidelity.

Audio fidelity
A key part of audio fidelity lies in the choice of amplifier. Most amplifiers are linear in nature, being either Class A, Class B or Class AB (see Figure 1 for a comparison of the three). While all linear amplifiers can deliver good gain with minimum noise, the signal-to-noise ratio (SNR) becomes more important in audio applications, due to the receiver (the human ear) being very good at discerning between the two. Each of the above classes of amplifier have their good points and their bad points when it comes to power and performance, hence the need for the different biasing classes.
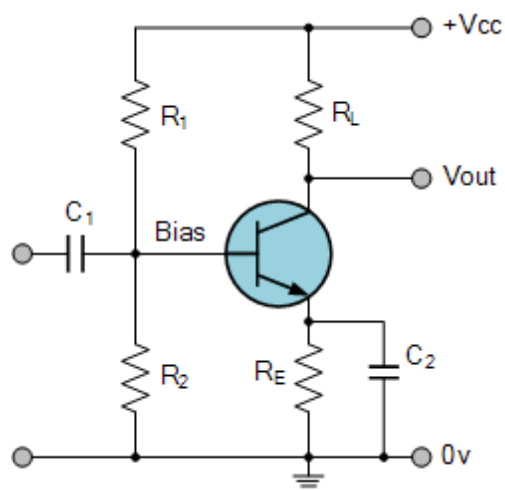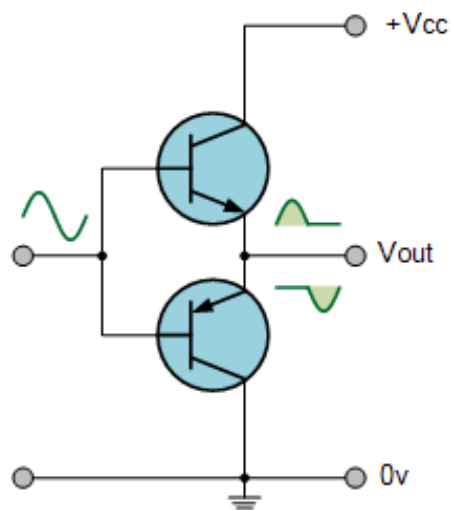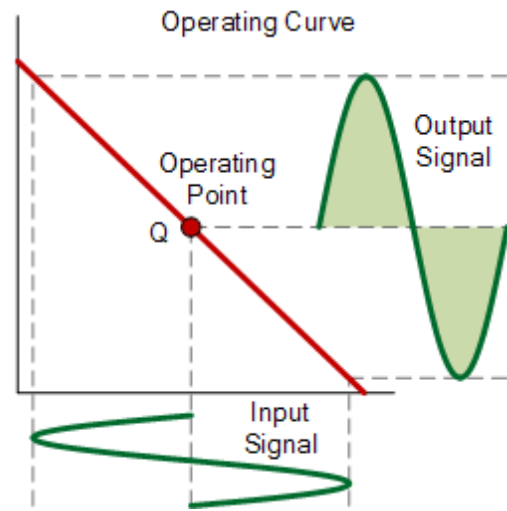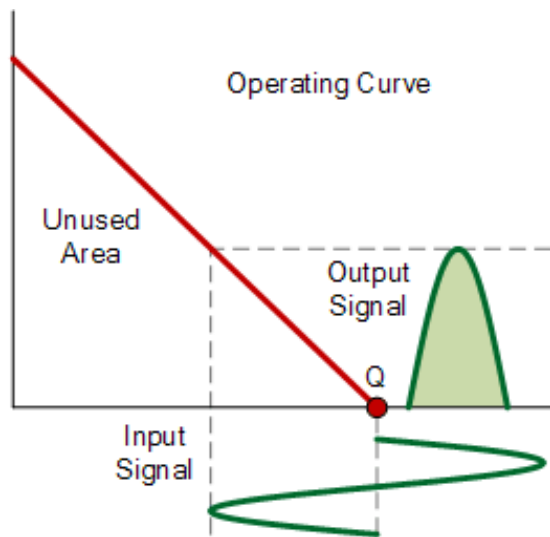
Figure 1a: A Class A amplifier
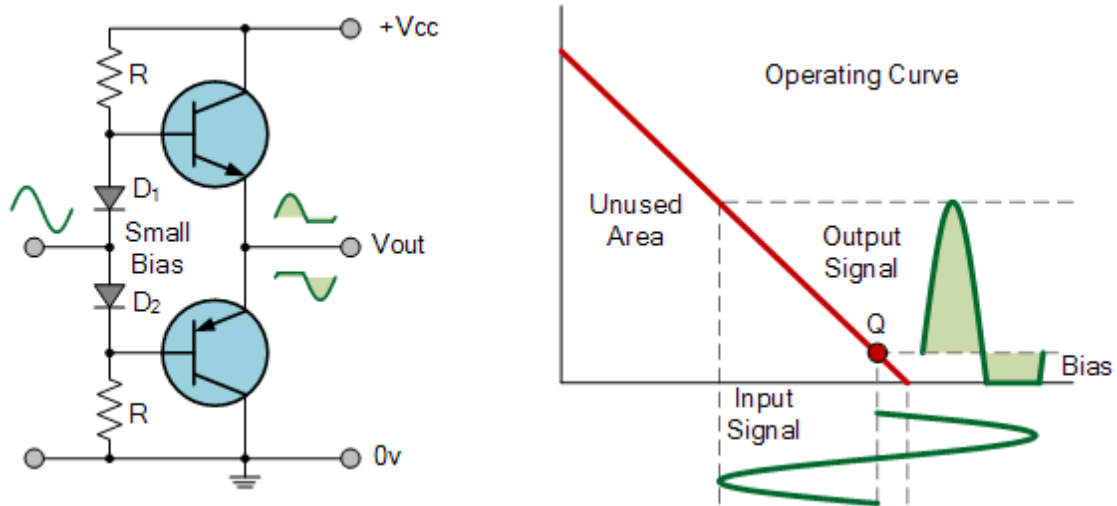

Figure 1b: A Class B amplifier

Figure 1c: A Class AB amplifier

In the kind of applications now emerging, efficiency both in terms of audio clarity and power will be important. All linear amplifiers are biased to be permanently 'on' to some degree, which delivers good linearity but poor power efficiency. For example, a Class A amplifier uses a single transistor biased such that it conducts even when the input signal is zero. Class B amplifiers use two transistors, one for each half-cycle, so in theory only one transistor is conducting at any one time. This can improve power efficiency but often at the cost of introducing crossover distortion. Class AB amplifiers improve on this by shifting the bias slightly to minimise crossover distortion. While Class AB is a good compromise in most cases, it can still only deliver around 80% efficiency, which for some battery-powered and portable applications may not be good enough.

The Class D amplifier is a significant departure from regular amplifier topologies. Instead of amplifying the input linearly, it digitises the signal, turning it into a pulse-width modulated (PWM) square-wave representation of the input signal. This 'fully on/fully off' approach can deliver very high efficiencies of over 90%, but comes with the additional need to filter the output in order to recover the analogue component of the signal, which can create additional design challenges. However, the PAM8014 Filterless Class D amplifier represents a significant advancement in its field, thanks to a number of design features. It offers 3.2 W of mono output power and an efficiency of

over 90%, with high power-supply rejection ratio and a differential design that can significantly reduce the impact of noise and RF reflections.

Making a better Class D amplifier

The Class D topology is typically realised using a comparator to compare the input signal with a reference signal to create what is effectively a 1-bit ADC; when the input signal is greater than the reference the comparator's output is high, and it is low when the input is lower than the reference, as shown in Figure 2.
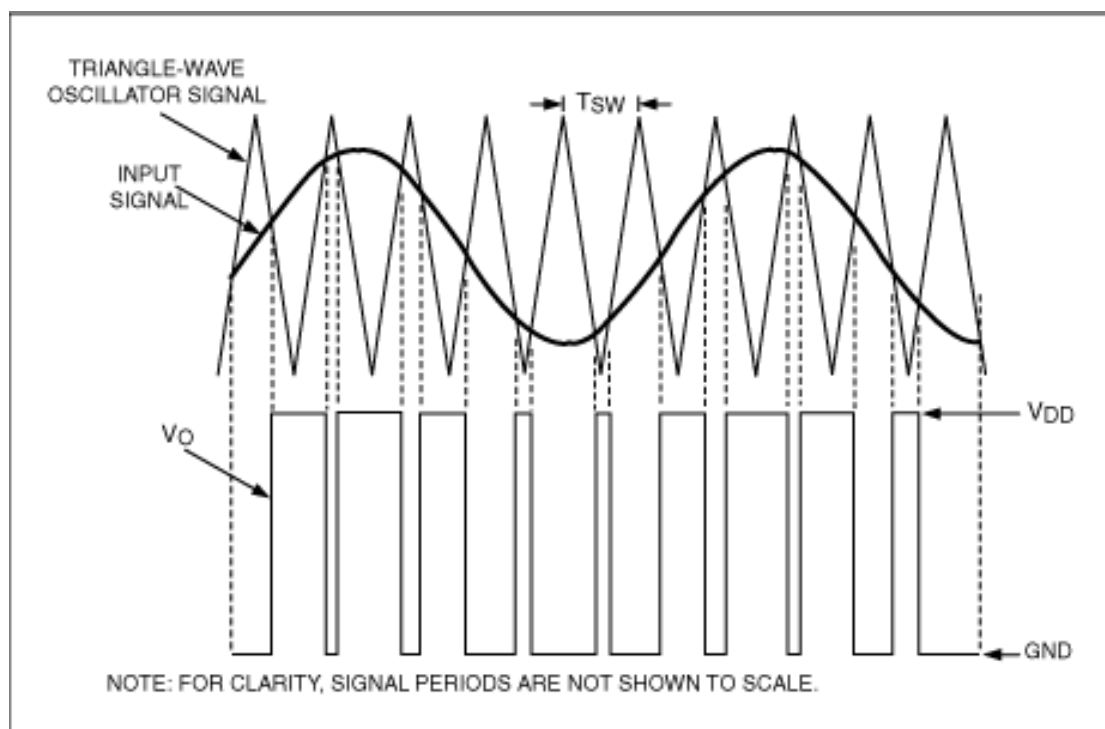
Figure 2: A Class D amplifier uses a comparator to turn an analogue signal into a series of pulses, creating a PWM output which normally needs to be passed through a low-pass filter in order to recover the analogue signal

The output is fed to the switching stage, formed by either two or four switching FETs. A two-FET output stage would typically be formed from complementary FETs (PNP and NPN), operating as a half-bridge. A disadvantage of this design is that current flows through the load even if the output is not changing, as the output is biased at $V_{DD}/2$. Using a full-bridge output stage overcomes

this. A full-bridge implementation uses four FETs, formed from two half-bridge stages. This produces a bidirectional swing in output current and is referred to as a bridge-tied load (BTL). Although this design uses twice as many FETs it provides significant benefits, not least because the offset is applied to both sides of the load, thereby removing the losses associated with the high quiescent current of a half-bridge design. Figure 3 shows a block diagram of the PAM8014 Class D amplifier, which features a full-bridge BTL design.
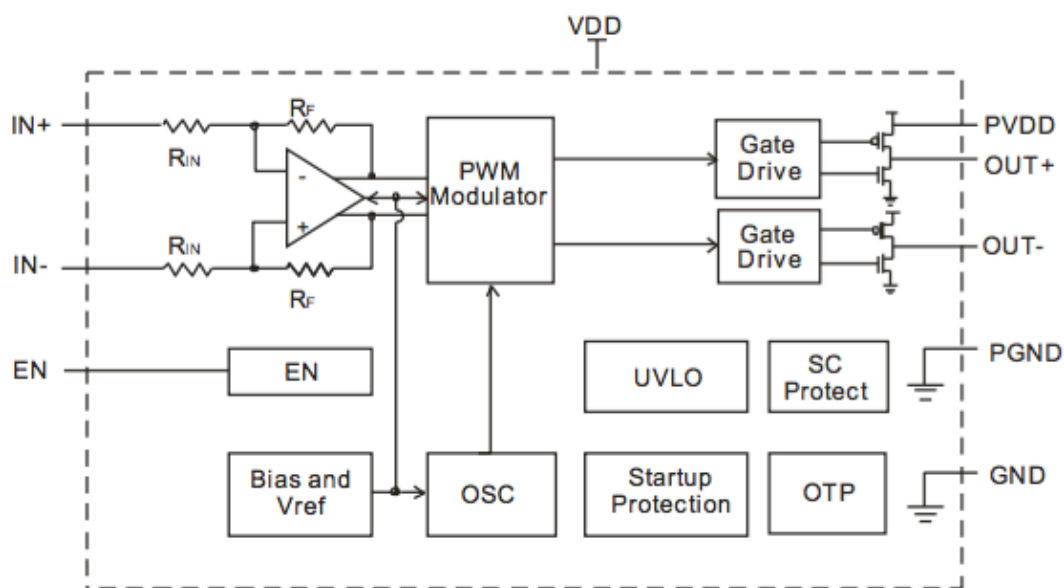
Figure 3: The PAM8014 is based on a full-bridge BTL topology to deliver superior performance in a tiny package

Integrating a BTL topology into a Class D amplifier requires careful design in order to achieve a useful level of output power in a single integrated package. In audio applications using more advanced speaker technology, this is less of an issue and actually opens up a host of new potential applications in portable and battery-powered devices.

No-pass filter
These system-level advantages might be lost if the Class D amplifier still requires the output stage to be passed through a large and potentially costly low-pass filter. The discrete passive components needed to construct a low-pass filter, such as an inductor big enough to handle the output power, can

easily take up more PCB space than the amplifier itself, as well as adding significant cost.

The low-pass filter is normally required to remove the high frequencies in the PWM output, so it is integral to many applications. The extra PCB space and costs required for the low-pass filter could mean that using a Class D amplifier isn't possible in some applications – however, this could actually be turned into a positive in audio applications.

Speakers are effectively formed from a coil, which is essentially an inductor with a series resistance; two of the components needed to realise a low-pass filter. The built-in inductance of a speaker can actually be exploited in this way; however, it does require the BTL output stage to have been designed to support filterless design. This is important because, if not, there could be a current flowing through the speaker at all times, which could result in a reduced operational lifetime. Also, if the output has a bias on it, it could limit the speaker's dynamic range. The BTL output stage of the PAM8014 Class D amplifier has been designed for filterless operation and so avoids all of these pitfalls, while also bringing the advantages of Class D operation to audio applications.

While the human ear is a natural high-frequency filter, it is also adept at picking up noise in all its forms. This is why a good Class D amplifier will implement internal feedback in order to deliver the best total harmonic distortion + noise (THD + N) figures; in this case the PAM8014 typically delivers 0.14% at an output power of 0.5 W @ 1 kHz.

Other advanced features of the PAM8014 include shutdown mode, which reduces the supply current drawn to a maximum of just 1 µA. The PAM8014 also integrates circuitry dedicated to minimising the 'pops and clicks' that can result from transients during turn-on and turn-off, or when coming out of shutdown mode. This is achieved using a special mode that mutes the internal amplifiers until the reference voltage is stable; once stable, full operating mode is restored. Other features include: under voltage lock-out

(UVLO), which puts the PAM8014 into shutdown mode if the supply voltage drops below 2.0 V; short-circuit protection (SCP), which protects the device should the outputs be shorted to each other or to ground; and over-temperature protection (OTP), which puts the device into shutdown mode if the die temperature rises above 150°C.

The PAM8014 is provided in the U-WLB1313-9 Wafer-Level BGA package (see Figure 4), an outline that measures less than1.3 mm on each side, making it suitable for even the smallest device.
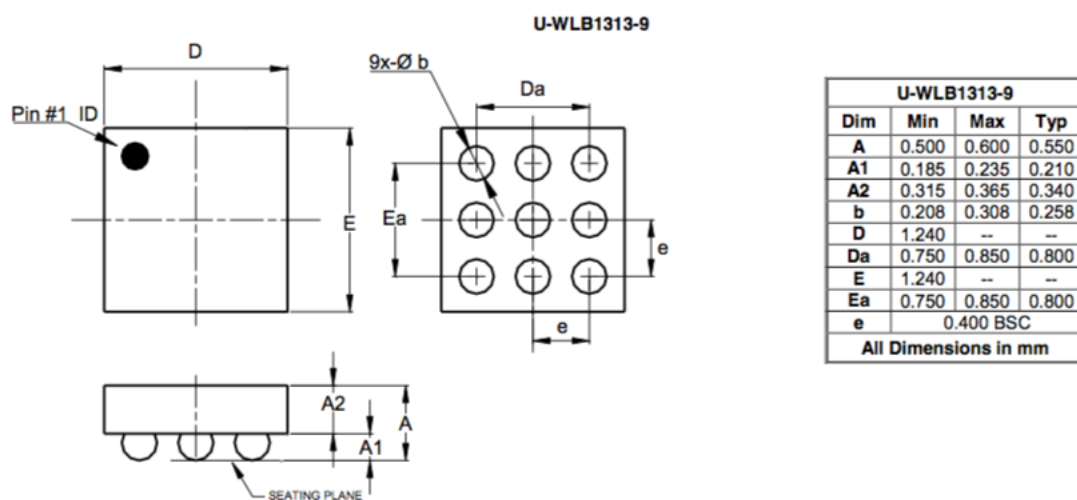
U-WLB1313-9

	U-WLB1313-9		
Dim	Min	Max	Typ
A	0.500	0.600	0.550
A1	0.185	0.235	0.210
A2	0.315	0.365	0.340
b	0.208	0.308	0.258
D	1.240	--	--
Da	0.750	0.850	0.800
E	1.240	--	--
Ea	0.750	0.850	0.800
e	0.400 BSC		
All Dimensions in mm			

Figure 4: The PAM8014 is provided in a Wafer-Level BGA package measuring less than 1.3 mm on each side

With a fixed gain of 18 dB and an efficiency of up to 93%, the PAM8014 requires a minimum of external components, as shown in Figure 5.
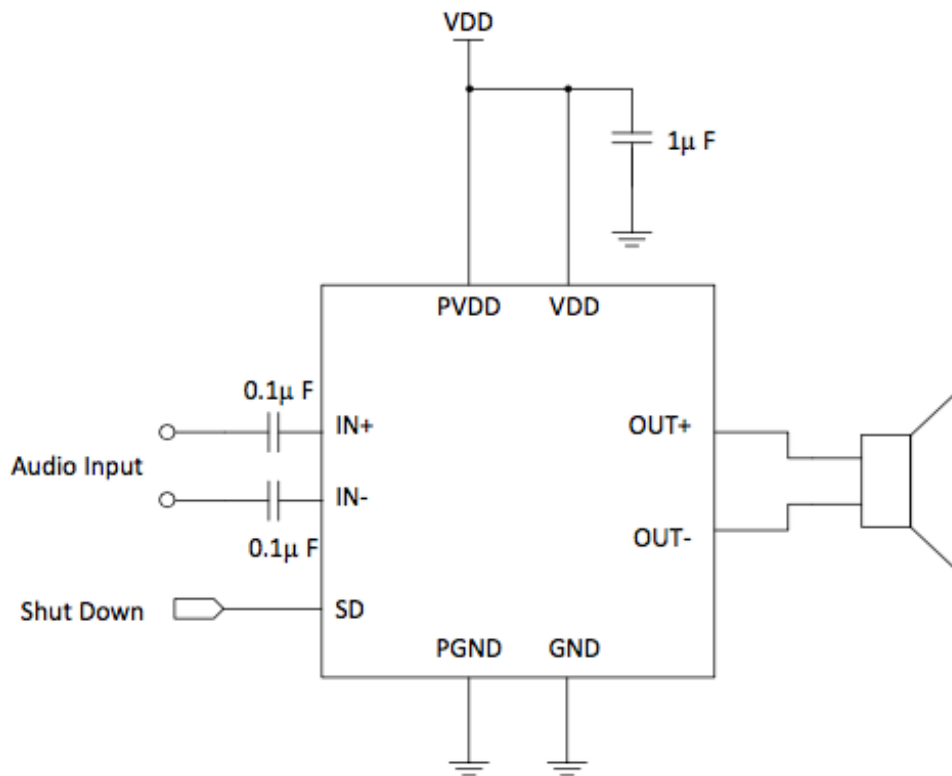
Figure 5: The PAM8014 requires very few external components, making it ideal for space-constrained and ultra-low power applications

These advanced features make the PAM8014 ideal for a wide range of applications, including smartphones, VOIP, MP4/MP3 players and, of course, digital assistants.