wolfson microelectronics

WAN_0140

ALC for Portable Applications

INTRODUCTION

In applications that offer a recording feature (such as a voice recorder), it can often be desirable to ensure that the recorded signal is kept at a constant level. For example, if recording voice, the signal may vary greatly as the user's volume changes and as the distance to the microphone changes. These effects can result in a recorded signal that is difficult to listen to when played back.

The purpose of the ALC is to maintain a consistent volume as the input signal level changes. In a typical CODEC, this is achieved by regularly adjusting the PGA gain in order to maintain the signal level at the ADC input at a specified target level. A Peak/Limiter and Noise Gate function are also frequently used in conjunction with an ALC; these functions are also described within this application note.

Setting up the ALC to be optimal for every different recorded source (such as voice, classical music, pop music etc) requires specifically tailored parameters. The aim of this document is to ensure that users of the ALC feature are fully aware of what its capabilities are and of the effects of changing particular parameters. Example settings for typical portable recording applications are also provided as a starting point for configuring an ALC.

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DEFINITIONS

ALC is frequently confused with other processes such as compression and peak limiting. The following paragraphs provide some definitions which will help to set these different functions apart and to see the purpose of ALC within an appropriate context.

DYNAMIC RANGE

Dynamic range of an audio signal is the ratio of the loudest signal level to the quietest signal level encountered. On a logarithmic (dB) scale, this corresponds to the distance in dB levels of the loudest and the quietest parts.

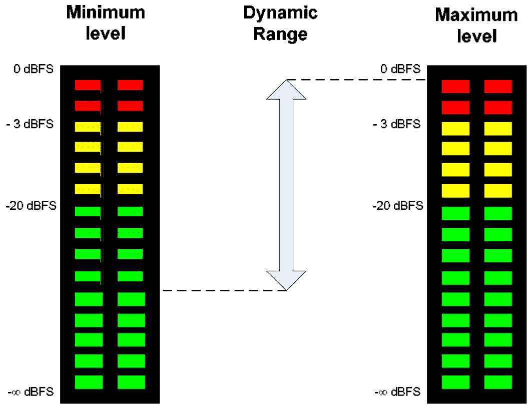


Figure 1 Dynamic Range

GAIN

Applying a linear gain to a signal adjusts the amplitude of every portion of the signal by an equal amount. This has the effect of lifting both the upper and the lower limits of the dynamic range; the dynamic range is not affected by this. Optimal adjustment of the gain enables the best possible signal to noise (SNR) performance to be achieved.

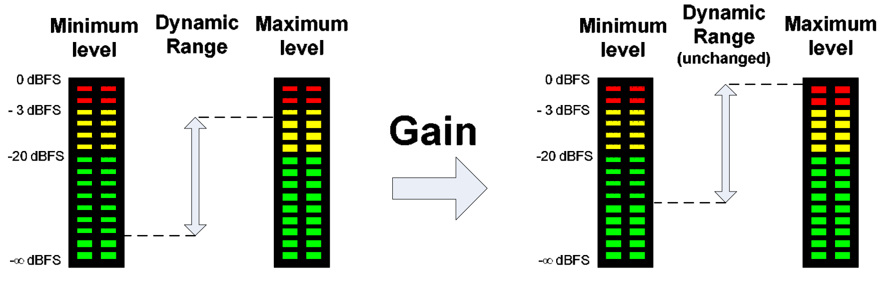


Figure 2 Gain

COMPRESSION

Applying compression to a signal adjusts the amplitude of the signal by a variable level that is a function of the RMS signal level. Low level signals are amplified more than high level signals. This has the effect of bringing the upper and lower limits of the dynamic range closer together, ie. The dynamic range has been reduced by this process. Compression can be used in order to adjust a signal for a transmission medium whose dynamic range is more limited than the original signal. Compression also has the effect of causing a transient signal to sound more 'punchy', as it increases the signal's energy without greatly altering its volume.

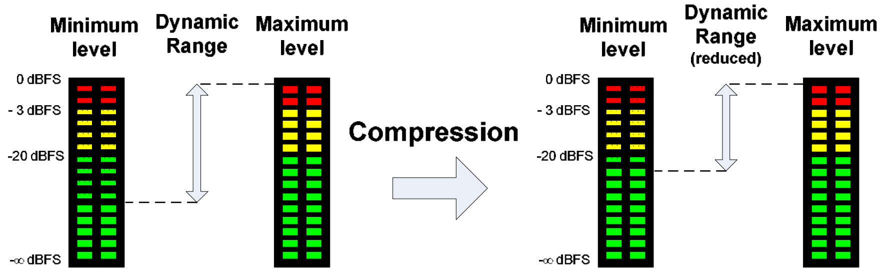


Figure 3 Compression

Note that, providing the compressor transfer characteristic is known, then the reverse process (known as dynamic expansion) may be applied in order to restore the original signal.

LIMITING

Limiting is a process designed to increase a system's tolerance to high signal levels. If a signal exceeds the maximum level that can be accommodated by an analogue system, clipping will occur at the maximum level, and the information contained in the clipped portion of the waveform is lost. This form of signal degradation is very noticeable to the ear and is almost universally undesirable.

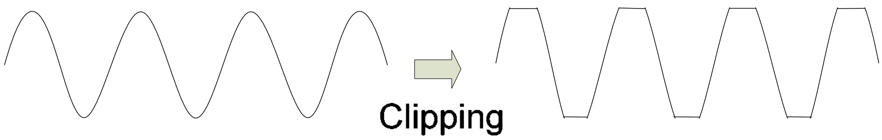


Figure 4 Clipping

If it is not possible to ensure that the maximum signal level will never be exceeded, signal limiting can be implemented by which the hard-edged waveforms are eliminated. This may be achieved by rapidly reducing the signal gain in the event of a peak signal level being reached. Alternatively, a more advanced approach may be adopted using digital processing to adjust the shape of the signal in order to preserve its features as far as possible without making unnecessary gain changes. This method introduces a small delay in the signal path in order that the processor can effectively 'look-ahead' at the signal that follows.

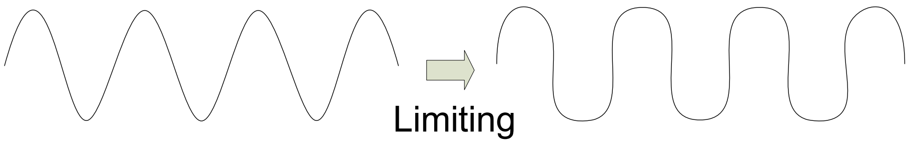


Figure 5 Limiting

AUTOMATIC LEVEL CONTROL

Automatic Level Control (ALC) is a function that controls the gain of a system adaptively. The gain is determined from the RMS level of the signal over a period of time. An ALC adjusts the signal gain in order to achieve a pre-determined target level and thereby prevent clipping and suppress any discontinuities between, say, different voices or different music tracks. ALC differs from compression by attempting to reduce the dynamic range to zero - i.e. produce a constant RMS output level - whereas compression would normally preserve more of the dynamic range of the processed signal.

If a signal level is consistently below the target level, the ALC will cause the gain to be increased until the target level is reached. If a signal level exceeds the target level, the ALC will decrease the gain. The time constants for these two actions are usually set independently: a fast response is usually required to avoid clipping in the event of a loud signal, whereas a slower response is more acceptable when increasing the gain in order that the gain adjustment is not too noticeable and that it does not excessively distort the dynamics of the signal source.

The following plots illustrate a short voice recording before and after ALC adjustment. Figure 6 shows the original signal.

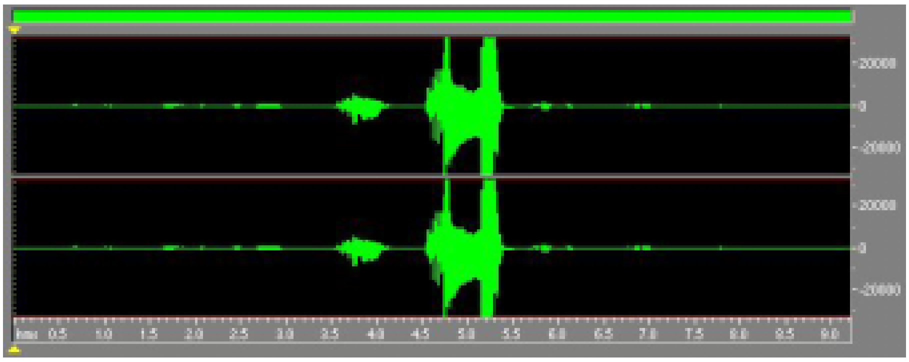


Figure 6 Voice Recorded File with no ALC Applied

The plot shown in Figure 7 shows the same audio file with ALC applied. It can be seen that the low level sections of the signal have been amplified, whilst the louder sections have been attenuated to avoid clipping.

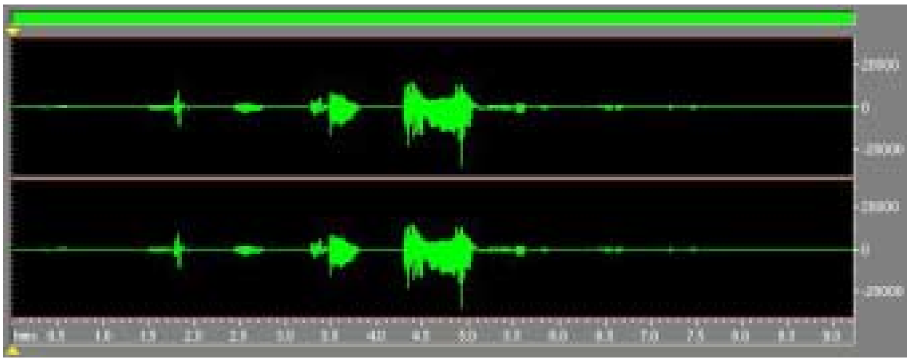


Figure 7 Voice Recorded File with ALC Applied

NOISE GATE

A Noise Gate is also a form of Automatic Level Control. Its function is to detect the absence of an input signal, or when the signal level falls below a given threshold and to control the signal level in an appropriate manner. Typically, the Noise Gate might be required to mute the signal when it falls below a given level; muting the signal will result in a quieter output than would be present if the signal noise were allowed to pass through the system. This function can be used effectively in conjunction with an ALC function in order to prevent the ALC from increasing the gain applied to background noise in an attempt to achieve the target signal level.

APPLICATIONS FOR AN AUTOMATIC LEVEL CONTROL

ALC has many different applications, each with slightly different aims. It follows, therefore, that an ALC needs to be set up specifically for an intended application and, if appropriate, re-adjusted for different applications.

SPEECH

When processing speech, an ALC can compensate for the movement of the speaker, or the different signal levels received from different parties of a conversation. An ALC can respond very quickly to the different signal levels and provide a more constant audio level that is more pleasing to listen to and likely to be more intelligible. The effect of ALC on a voice signal is illustrated in Figure 6 and Figure 7.

MUSIC

When processing music, in some applications it can be desirable to ensure a constant volume level is maintained. This can avoid the need to manually adjust the volume between one signal source and another, or between consecutive music tracks whose levels are imbalanced. Care must be taken, however, that musical detail in the audio signal is not totally destroyed by the ALC adjustment of the volume level; the ALC must preserve the dynamics of the musical signal to a much greater extent than it would with a speech signal. The effect of ALC on a musical signal is illustrated in Figure 8 and Figure 9.

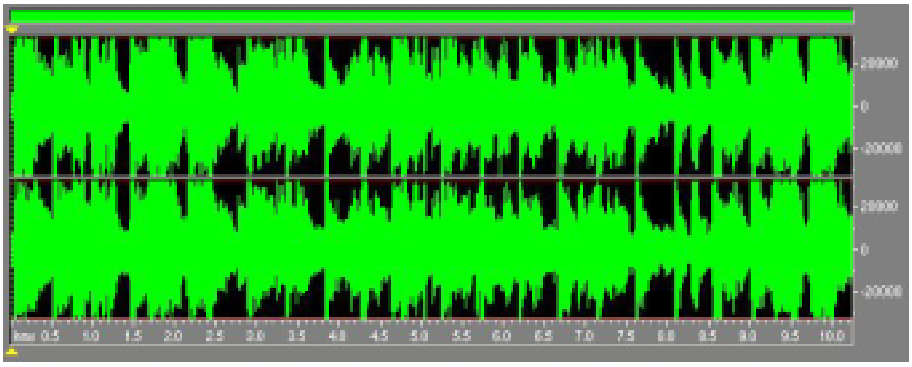


Figure 8 Audio Music File with no ALC Applied

The plot shown in Figure 9 shows the same audio file with ALC applied. It can be seen that the signal level has been reduced to the desired ALC level, which has been set in order to avoid any clipping of loud signals.

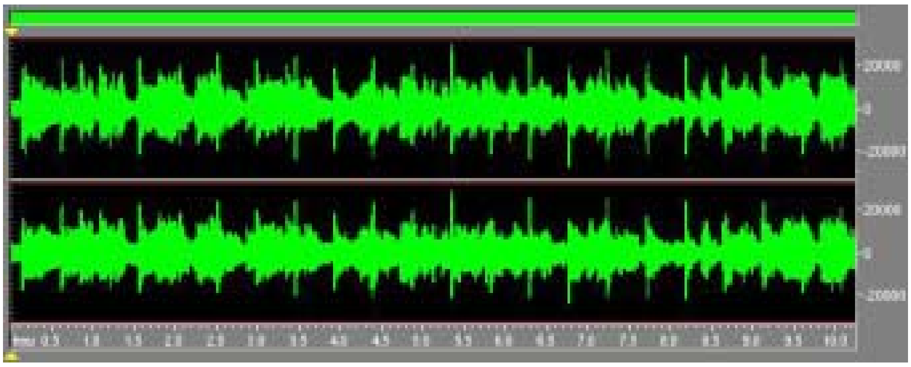


Figure 9 Audio Music File with ALC Applied

CONTROL FIELDS

The ALC functions in Wolfson CODECs are highly adaptable on account of the number of different parameters that may be individually set. The following paragraphs describe each of these parameters, and also some of the constraints or tradeoffs that determine the optimum setting for each. As described earlier, different parameter values will be desirable to suit different types of audio signal. Personal preferences can also influence the choice of settings.

AUTOMATIC LEVEL CONTROL

The Automatic Level Control operation is set by a number of parameters in the time domain and in the amplitude domain. Figure 10 illustrates the ALC operation in adjusting the gain over a period of time in order to maintain the ALC target level.

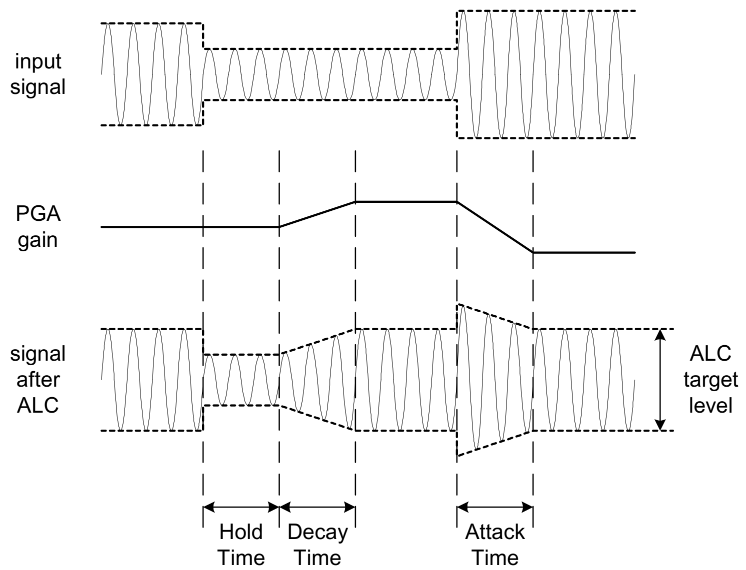


Figure 10 ALC Operation

ALC Target Level is the desired signal level. In a CODEC, this level would be set below the ADC full-scale level in order to reduce the likelihood of clipping in the event that the input signal level increases. This level should be set as high as possible in order to achieve the best signal to noise performance, but not so high as to allow signal clipping to occur as the signal changes. The more erratic the signal level, the greater the required headroom between the ALC Target Level and the ADC full-scale level.

Hold Time is the time delay between a low signal level being detected and the gain starting to be increased. The purpose of this delay is to ensure that the ALC is not over-responsive to a changing signal level. If the signal level rises again before the Hold Time has expired, then no gain adjustment is made and the dynamics of the original signal are preserved. The Hold Time should be set according to the type of ALC response that is desired. A short Hold Time should be used if an immediate gain adjustment is required to a changing signal; this might be applicable to voice applications. A longer Hold Time should be used if the ALC gain adjustments are to be made more sparingly, thus responding only to long term signal level changes and preserving the original signal dynamics to a greater extent; this might be applicable to music containing a large dynamic range, as is frequently found in classical music.

Decay Time is a measure of the rate at which the gain can be increased by the ALC. It can be represented in many different ways, such as the time taken to ramp up across 90% of its range, or the time taken to ramp up 6dB, or the time per gain step. All these representations are directly related to each other. The Decay Time determines how rapidly the ALC will make adjustments in response to a fall in signal level. Note that the actual time taken to adjust the gain will depend upon the extent of the gain adjustment required. A short Decay Time should be used if a fast response is required to a changing signal. The Decay Time should not be so short as to cause rapid ALC response to a nominally constant signal level. For example, if the input signal is likely to have pauses or silences, the Decay Time should be set long enough to ensure that the ALC is prevented from making large adjustments to the gain during those durations.

Attack Time is a measure of the rate at which the gain can be decreased by the ALC. It is measured similarly to the Decay Time. As before, the actual time taken for the ALC to adjust the gain is a function of this parameter and also the extent of the gain adjustment required. The advantage of a short Attack Time is that it results in a fast response to an increased signal level. This in turn reduces the possibility of clipping, as the signal is quickly shifted away from the clipping level. Many of the same considerations apply as for Hold Time and Decay Time. The Attack Time should be set in conjunction with the Decay Time. For instance, it should be considered that, if the system responds rapidly to a short burst of increased signal level, and then responds slowly to restore the level again afterwards, this may be an undesirable characteristic. On the other hand, if a more steady input signal is anticipated, then a slower Attack/Decay Time may be most suitable.

For further optimisation of the ALC function, the user can set limits to the amount of gain that the ALC function may select.

Minimum Gain is the minimum amount of amplification that can be commanded under control of the ALC function. The purpose of the minimum gain is to ensure that large input signals are permitted and not excessively attenuated by the ALC function. If the Minimum Gain is large, then the ALC will be restricted in its ability to control the signal level and there is a greater possibility that it will be unable to prevent distortion of large signals. However, if the Minimum Gain is small, then a greater attenuation will be applied to large signals, which may undesirably limit the dynamic range of the processed signal. The Minimum Gain should be set as low as is possible, and certainly no greater than the gain that would be required to adjust the largest input signal down to the ALC target level.

Maximum Gain is the maximum amount of signal amplification that can be commanded under control of the ALC function. The purpose of the maximum gain is to ensure the small input signals are accommodated and not excessively amplified by the ALC function. For example, if a recorded music track fades out; the Maximum Gain setting prevents ALC from destroying the effect by continually increasing the gain as the music signal fades. If the Maximum Gain is large, then the ALC is more likely to over-compensate for a fading signal. If the Maximum Gain is small, then the ALC will be restricted in its ability to control the signal level. The Maximum Gain should be determined from the level of a quiet signal that the designer determines should be treated as a fading signal. For example, if the ALC Target Level is -10dB and the designer determines that any signal below -40dB should be allowed to fade, then the Maximum Gain should be set at 30dB.

Setting up the ALC requires an understanding of the type of signal that requires to be accommodated. It is not possible for a single ALC configuration to respond optimally to all types of input, as every scenario would require different settings. It must be understood that compromises will be necessary if a single ALC configuration is required to accommodate every possible kind of input.

If an ALC is configured with slow response characteristics suitable for music, then it will not respond optimally to fast impulses. Equally, if an ALC is configured to handle accommodate fast transient sounds (such as a handclap), then undesirable gain pumping effects may be heard when processing less transient signals. Therefore, the ALC configuration must consider all types of sounds that require to be accommodated and build in appropriate compromises.

It should be noted that the ALC is capable of responding to sounds that the listener is not aware of - this is sometimes evident when there is very low frequency content in the signal. In this event, it may be beneficial to increase the Hold, Attack and Decay times in order to decrease the ALC's responsiveness. Incorrect behaviour can also arise from DC offsets in the audio signal path; this can cause the ALC to incorrectly measure the signal level, leading to incorrect gain adjustments being made.

PEAK/OVERLOAD LIMITER

The Peak/Overload limiter is used in conjunction with the ALC function. It is designed to ensure that no clipping of the processed signal occurs. The signal is particularly prone to clipping in the event of a loud signal occurring after a quiet period for which the ALC has increased the signal gain.

The Peak Limiter function overrides the normal ALC settings in the event of the signal output exceeding the Peak threshold (fixed at 1.16dB below full scale). The gain is decreased at the maximum rate (overriding the 'Attack Time' setting) until the signal is below the threshold. The Peak Limiter function is always enabled whenever the ALC is enabled in normal (ALC) mode. There are no user-selectable parameters associated with it.

On some Wolfson ALCs, the Limiter can be selected as a specific type of ALC operation. This Limiter operation is very similar to normal ALC, except that the maximum ALC gain is set equal to the PGA gain at the time when Limiter mode was selected. As a result, the function will reduce the signal gain in response to peaks, but will never boost the signal with respect to the start-up level. This type of Limiter function may be selected in place of the ALC function; it is not possible to select both modes simultaneously. The same user parameters apply in Limiter mode as in ALC mode, though the implementation permits shorter Attack/Decay times in Limiter mode.

NOISE GATE

The noise gate feature is also a part of the ALC function and is designed to control the signal chain in the absence of a signal, or when the signal level falls below a given threshold.

When the signal is very quiet and consists mainly of noise, the ALC function may cause 'noise pumping', which occurs during quiet periods as the gain is increased in an attempt to achieve the target signal level. The increased noise level is noticeable and highly undesirable. The noise gate function helps to prevent this noise pumping by comparing the pre-ALC signal level with a noise gate threshold, NGTH.

When the Noise Gate Threshold is exceeded, then either the signal will be muted, or else the gain will be held constant (preventing it from ramping up as it normally would when the signal is quiet). The choice between these two actions is determined by the user, as described below.

Noise Gate Threshold is used to set the noise gate threshold with respect to the ADC full-scale level. This is the signal level below which the noise gate takes effect. Care must be taken to set an appropriate threshold; levels at the extremes of the range are likely to cause noticeable and inappropriate operation. If the threshold is too high, then the noise gate will activate too readily and a large amount of the signal content may be lost. If the threshold is too low, then the noise gate will rarely be activated and noise pumping is more likely to occur.

Noise Gate Type is used to select what will happen when the signal falls below the threshold level. There are two options:

- **Gain held constant** causes the gain setting to remain fixed at the level it was at prior to exceeding the noise gate threshold. This setting is appropriate if the gate is to be prevented from muting quiet signals that may be a desired feature of the signal source.
- **Output mute** causes the output to be clamped to ground when the noise gate threshold is exceeded. This will offer the best noise reduction but may undesirably distort the input signal in quiet moments. This setting is more appropriate for signals where fidelity is not important and the ALC can reliably identify silences in the input signal.

The two types of Noise Gate are illustrated below. Figure 13 shows a musical source signal, without any Noise Gate processing.

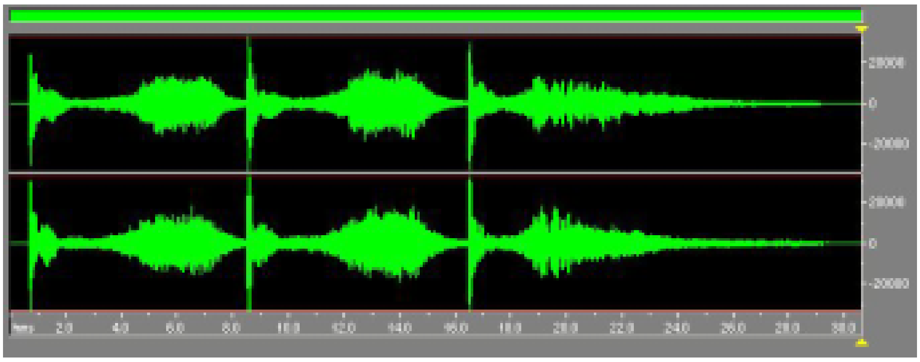
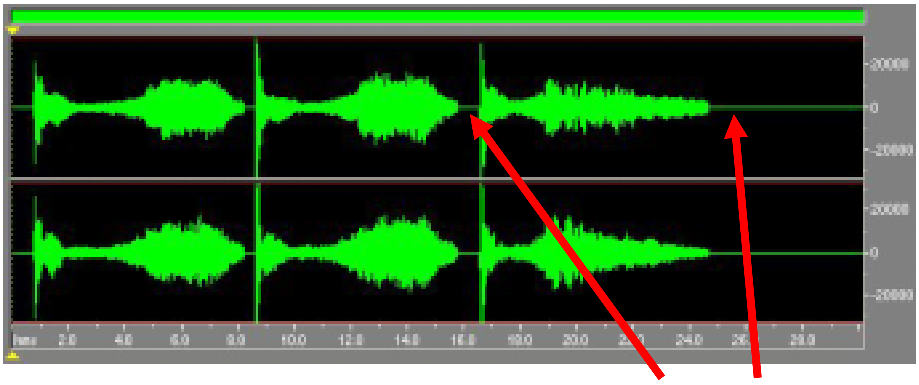


Figure 11 Music File with no Noise Gate

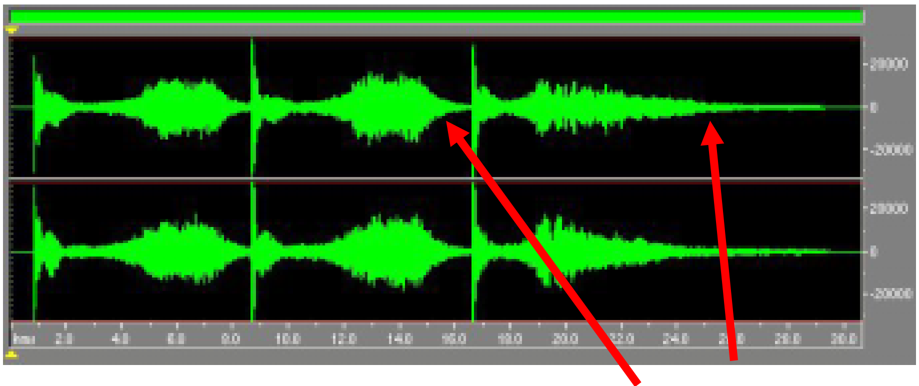
Figure 12 shows the source signal again, with the Noise Gate function enabled and the "Output Mute" option selected. It can be seen that this has resulted in the quiet sections of the music being muted, giving a poor representation of the original piece.



Quiet sections muted by Noise Gate

Figure 12 Music File with Noise Gate Enabled and "Output Mute" Option Selected

Figure 13 shows the same source signal, but the noise gate is now enabled with the "Gain Held Constant" option selected. It can be seen that there is no obvious difference between this and Figure 11. Thus, the dynamics of the original signal have been preserved.



Quiet sections not affected by Noise Gate

Figure 13 Music File with Noise Gate Enabled and "Gain Held Constant" Option Selected

MISCELLANEOUS ALC OPTIONS

Additional control fields relating to the Wolfson ALCs are described in the following paragraphs.

ALC Select is the parameter that determines which of the CODEC's audio channels will be subject to ALC control. The available options are

- ALC Off - neither channel is under ALC control; both are controlled by the PGA gain register settings
- Right Channel only - only the Right channel is under ALC control; the Left channel is under fixed register control
- Left Channel only - only the Left channel is under ALC control; the Right channel is under fixed register control
- Stereo ALC - both Left and Right channels are under ALC control.

The Stereo ALC configuration is illustrated in Figure 14.

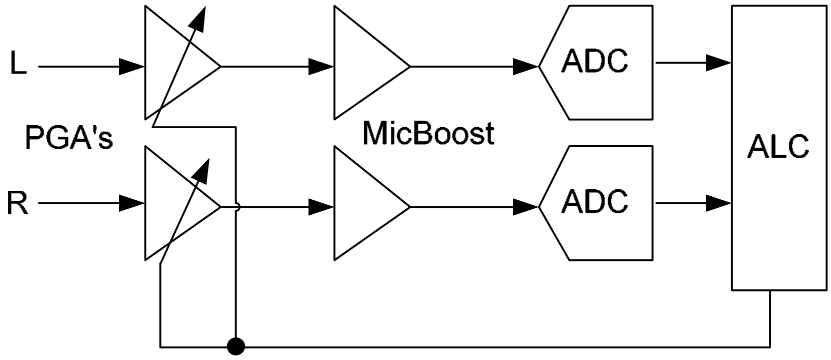


Figure 14 ALC Block Diagram

When ALC is selected, the start-up setting is read from the applicable PGA gain register. The gain thereafter will be controlled in both channels simultaneously, but the relative balance between the two will be unchanged. If, for example the Left and Right channels were set to +0dB and +10dB respectively at the moment when Stereo ALC is selected, then the 10dB difference would be retained for as long as the ALC is active. In most applications, therefore, it is advisable to ensure that the Left and Right PGA settings are set to a common value prior to selecting Stereo ALC mode.

In some CODECs, the Stereo ALC function is controlled by the signal level in the Left channel only. This can result in poor ALC performance in situations where the audio signal differs greatly between the Left and Right channels.

Note also that when the ALC feature is set active, any updates to the input PGA settings will be ignored. The ALC overrides the PGA volume, the channel mute and the zero-cross parameters. The latest PGA register values will be restored when the ALC function is switched off.

ALC Zero-Cross is a feature that is used to reduce the effect of 'zipper noise' when the PGA gain is updated. When the ALC Zero-Cross function is enabled, the PGA gain setting will only be updated when the input signal crosses the zero level (normally, this equates to AVDD/2). When the ALC Zero-Cross is disabled, the PGA gain setting will be updated at the earliest opportunity, and may result in audible clicks arising from the sharp edged discontinuities in the audio waveform. On some ALCs, the PGA gain steps are small enough to eliminate the need for Zero-Cross detection. For this reason, the Zero-Cross option is not required by some ALC designs.

Note that the Zero-Cross function can fail if the input signal does not cross the zero level for any reason. This can occur particularly if a DC offset is present on the signal. A timeout is provided to ensure that the gain may still be updated if a zero-cross has not occurred within a fixed time. The timeout is enabled via a register setting; it is not automatically enabled.

ALC Mode is the parameter that determines whether either the ALC Mode or the Limiter Mode is enabled. The Limiter mode rapidly adjusts the signal gain to prevent clipping, but does not boost quieter signals by any more than the initial gain setting. The ALC Mode adjusts the signal gain to a greater extent in attempting to maintain a desired target level. In many respects, the Limiter Mode can be considered to be a preset configuration of the ALC.

ALC Sample Rate is a parameter that requires to be set according to the ADC sample rate in order that the CODEC can correctly implement the ALC time controls. Failure to set this field correctly will result in the Hold, Decay and Attack times being implemented incorrectly.

AUTOMATIC LEVEL CONTROL AND NOISE GATE OPERATION

Figure 15 illustrates the combined functionality of the ALC and the Noise Gate under steady state conditions. The following features can be observed:

- There is a Noise Gate threshold, below which the input signal is attenuated by more than the ALC Minimum Gain.
- As the input signal level increases, the ALC Maximum Gain is applied, under which the output signal still falls short of the ALC Target Level.
- There is a range of input signal levels for which the ALC output will be at the ALC Target Level.
- When the input signal level increases even further, the signal is boosted by the ALC Minimum Gain only, and the output level approaches the 0dB output level.
- As the input signal level increases further still, the ALC Limiter attempts to prevent signal clipping at the maximum 0dB output level.

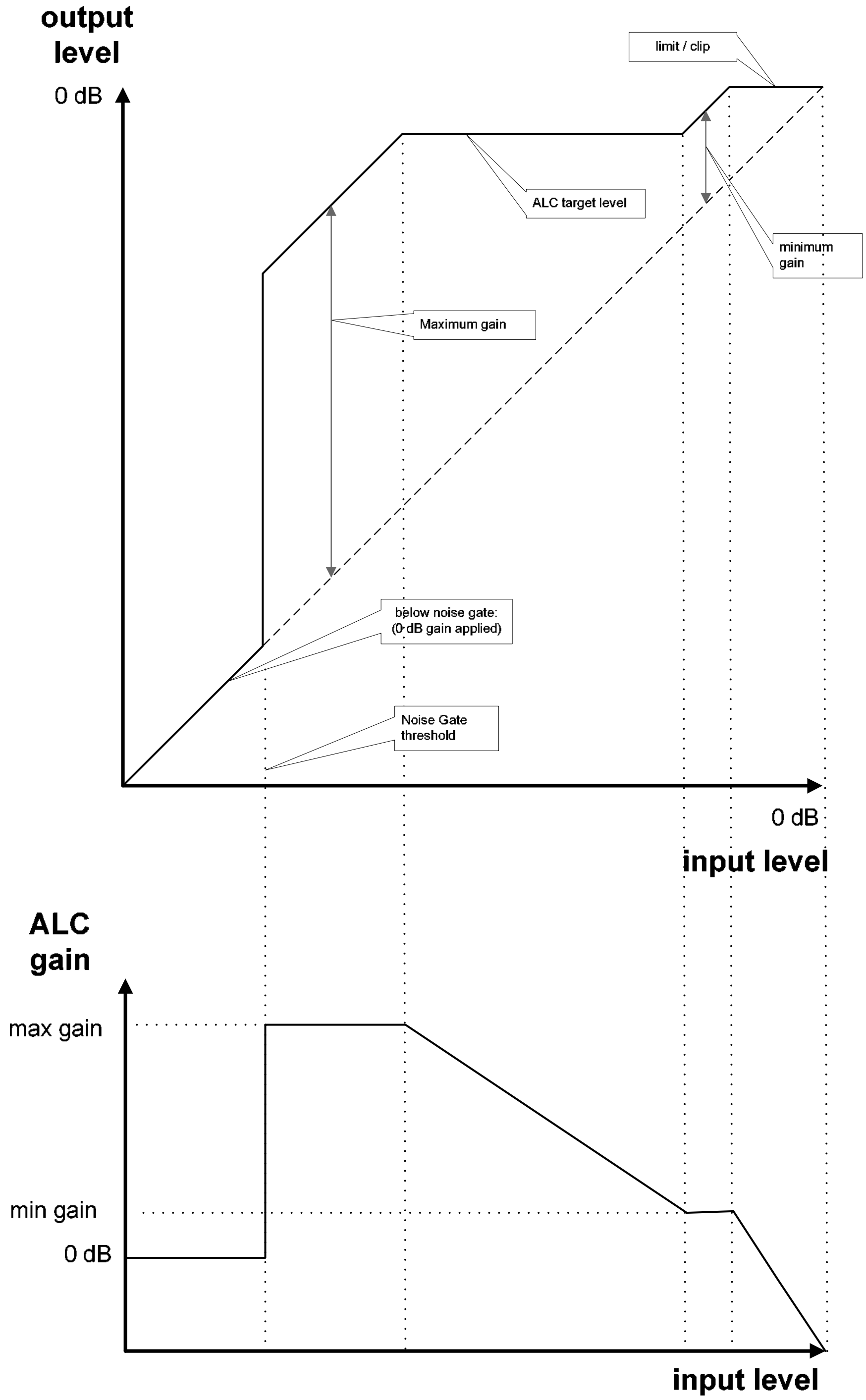


Figure 15 Amplitude and Gain Functions of an ALC (steady state)

RECOMMENDED SETTINGS

Recommended settings are provided below for a number of typical portable recording applications. These include speech recording and music recording. A generic setting is also provided, which aims to cater for the widest possible range of sounds.

It is important to note that these are suggested initial values only, as a starting point from which to derive the best settings for a particular circuit application. The quoted settings should give adequate performance in many cases, but it may be possible to improve the ALC performance through further adjustment of these settings.

Different products within Wolfson's range of CODECs offer slightly different ALC controls, but the principles of operation are comparable in all cases. Register settings for the WM8960 ALC are quoted for reference; Designers should consult the applicable product datasheet to confirm the required register values for the intended CODEC device.

SPEECH

For speech recording, a fast ALC response is desirable in order to quickly compensate for different people's voices, movement relative to the microphone. Failure to respond quickly may result in prolonged clipping of loud signals and a failure to adequately record quiet signals.

PARAMETER NAME	SETTING	WM8960 REGISTER	WM8960 VALUE
ALC Select	Stereo	ALCSEL [1:0]	11
Minimum Gain	-17.25 dBFS	MINGAIN [2:0]	000
Maximum Gain	+30 dBFS	MAXGAIN [2:0]	111
ALC Level	-12 dBFS	ALCL [3:0]	0111
ALC Zero-Cross	Off	N/A	N/A
ALC / Limiter Mode	ALC Mode	ALCMODE	0
Hold Time	0 ms	HLD [3:0]	0000
Decay Time	384 ms	DCY [3:0]	0100
Attack Time	24 ms	ATK [3:0]	0010
Sample Rate	*(see note below)*	ADC_ALC_SR	*(see note below)*
Noise Gate Enable	Enabled	NGAT	1
Noise Gate Threshold	-76.5 dBFS	NGTH [4:0]	00000
Noise Gate Type	Gain held constant	N/A	N/A

Table 1 Recommended Speech Record Settings

Note: The Sample Rate parameter, ADC_ALC_SR, must be determined from the particular circuit application; it is not possible to provide meaningful recommended settings.

MUSIC

For music recording, a slower ALC response is desirable in order to preserve more of the dynamics of the original signal. The ALC zero-cross option is enabled in order to avoid clicks being heard when the gain is adjusted.

A higher ALC target level than for speech recording may be appropriate here if the signal is less likely to contain unexpected transient peaks, and thus the ALC will be less prone to clipping. Also, a reduction in the maximum gain setting may help to avoid clipping when the music level increases after a quiet period and to restrict the extent of the ALC adjustments. Note that these adjustments may not be desirable in all music applications and are therefore not shown in the recommended settings. These are just some of the many adjustments that the user should consider when optimizing for a known operational environment.

PARAMETER NAME	SETTING	WM8960 REGISTER	WM8960 VALUE
ALC Select	Stereo	ALCSEL [1:0]	11
Minimum Gain	-17.25 dBFS	MINGAIN [2:0]	000
Maximum Gain	+30 dBFS	MAXGAIN [2:0]	111
ALC Level	-12 dBFS	ALCL [3:0]	0111
ALC Zero-Cross	On	N/A	N/A
ALC / Limiter Mode	ALC Mode	ALCMODE	0
Hold Time	0 ms	HLD [3:0]	0000
Decay Time	24.58 s	DCY [3:0]	1010
Attack Time	384 ms	ATK [3:0]	0110
Sample Rate	*(see note below)*	ADC_ALC_SR	*(see note below)*
Noise Gate Enable	Enabled	NGAT	1
Noise Gate Threshold	-76.5 dBFS	NGTH [4:0]	00000
Noise Gate Type	Gain held constant	N/A	N/A

Table 2 Recommended Music Record Settings

Note: The Sample Rate parameter, ADC_ALC_SR, must be determined from the particular circuit application; it is not possible to provide meaningful recommended settings.

GENERIC

For applications where the nature of the audio signal is unknown and the ALC must attempt to accommodate all types of sounds, a compromise setting must be found.

When the ALC is optimized for a different type of signal to the signal that is applied, poor performance may be observed. An example of this is the ALC's response to an impulse, such as a handclap. If the ALC Attack Time is short, then the ALC will respond to the handclap by rapidly reducing the PGA gain, with the possibility that the following signal will be too heavily attenuated. This will be particularly noticeable if the ALC Decay Time is slow. For an ALC response that is tolerant to impulses such as handclaps, it is recommended that the ALC Attack time should not be set below 192ms and the ALC Decay time should not be set above 192ms.

The minimum and maximum gain settings could be adjusted to restrict the extent of the ALC control if desired. The combination of settings should allow the ALC to respond quickly to changes in signal level and to impulse-type sounds, but also to minimise gain pumping caused by the associated level changes.

PARAMETER NAME	SETTING	WM8960 REGISTER	WM8960 VALUE
ALC Select	Stereo	ALCSEL [1:0]	11
Minimum Gain	-17.25dB	MINGAIN [2:0]	000
Maximum Gain	+30dB	MAXGAIN [2:0]	111
ALC Level	-12dB	ALCL [3:0]	0111
ALC Zero-Cross	On	N/A	N/A
ALC / Limiter Mode	ALC Mode	ALCMODE	0
Hold Time	0 ms	HLD [3:0]	0000
Decay Time	192 ms	DCY [3:0]	0011
Attack Time	192 ms	ATK [3:0]	0101
Sample Rate	*(see note below)*	ADC_ALC_SR	*(see note below)*
Noise Gate Enable	Enabled	NGAT	1
Noise Gate Threshold	-76.5dB	NGTH [4:0]	00000
Noise Gate Type	Gain held constant	N/A	N/A

Table 3 Recommended Generic Record Settings

Note: The Sample Rate parameter, ADC_ALC_SR, must be determined from the particular circuit application; it is not possible to provide meaningful recommended settings.

The example settings have been determined by listening tests and are fairly subjective; different users may have other preferences, and the settings may perform differently with varying sound sources. Nonetheless, these settings should provide acceptable performance in most cases and are a base to work from to optimise a specific application.

DESIGN TIPS

Care must be taken not to optimise too specifically for one type of signal source if the ALC will be expected to handle a wider range of signal types. For example, the ALC should not be optimised for slow-changing music and expected to accommodate a handclap perfectly. Equally, the ALC should not be optimised for speech if it will also be expected to handle all types of music.

Designers should consider whether there is any scope to reduce the demands on the ALC through knowledge of the likely signal type. For example, if the audio source is a Line Input, then it is more likely that the source is music whilst if the audio source is a Microphone Input, then the source may be either speech or music. Different ALC settings could be applied intelligently according to this information.

If there are any predictable sources of interference or other sounds that will affect the ALC operation, designers should consider whether the ALC can be disabled or muted at the time of the associated events. For example, the mechanical noise of a button-press near to the microphone may result in an undesirable response from the ALC. In some applications, it may be possible to mute the signal or disable the ALC until after the “Start Record” button has been pressed, for example. This is particularly relevant to portable applications, as the small size of the equipment is more likely to result in the push buttons being close to the microphone.

TESTING

The following paragraphs provide some guidelines as to how to verify the correct operation of the ALC. Note that this does not constitute optimisation of the settings, which must be largely determined by subjective judgement.

As the ALC function responds dynamically to the input signal, it can be difficult to test its functionality. It is recommended to test the steady-state gain levels separately from the time-domain parameters.

To test the steady-state gain levels, use an input signal with a frequency many times higher than (1/ATK). This will ensure that the ALC responds to the amplitude of the wave and not to the oscillating instantaneous amplitude of the signal. Whenever the input signal amplitude is changed, allow the ALC to fully settle before measuring the output signal. The settling time will be HLD+DCY in the case of a decrease in input signal, or ATK in the case of an increasing input signal. Under test conditions, it should be possible to verify the steady-state gain transfer function illustrated in Figure 15.

To test the time-domain response, use an input signal whose amplitude can be instantaneously stepped between the test levels. A tone burst that switches between two different amplitudes is ideal, providing that the period of the burst pattern is much larger than MAX ((HLD+DCY), ATK).

TROUBLESHOOTING

SYMPTOM	CAUSE	REMEDY
Gain pumping	Attack, hold and decay times are very short. The ALC responds too quickly to dynamics as a result	Set the attack, hold and decay times to longer values.
	The input signal is extremely dynamic in nature so that the gain travel is very large	Set mingain and maxgain to levels closer to the average gain to reduce gain travel
	The input signal contains large silences and the ALC is amplifying the noise during these silences	Activate the noise gate function or set the threshold value to a higher level
Output level is too loud	Target level is too high	Reduce target level
	Mingain is lifting gain above target level	Reduce mingain
	Signal is too dynamic to be caught by the ALC. This can be either the result of too long attack times or a very high noise gate level.	Reduce attack time Reduce noise gate level
Output level is too quiet	Target level is too low	Increase target level
	Maxgain is limiting the gain	Increase maxgain
	Decay time is very high and the level takes very long to increase, resulting in too low average level	Reduce hold time Reduce decay time
ALC responds to inaudible sounds	The signal contains low frequencies that the listener is not sensitive to.	Set the attack, hold and decay times to longer values.
	There are filters after the output of the ALC. The ALC should be the final part of the signal chain.	Disable any filters after the ALC. Any required filtering should be applied before the ALC.
ALC fails to make any gain adjustments	In Stereo ALC mode, the PGA gain adjustments only become effective when latched on both channels. This cannot be achieved if one or other ADC is powered down.	If one or other ADC is powered down, ensure that the ALC is enabled on the active channel only.
Noise gate is not functioning at the correct threshold.	DC offset in signal path is causing the ALC to measure the signal level wrongly.	Amend signal path to reduce DC offset Increase noise gate threshold
Output signal is muted	PGA gain was in its default state (mute) when the ALC and Noise Gate are initialised. The muted input signal results in the Noise Gate being permanently triggered, regardless of input signal.	Ensure that the input PGA is un-muted prior to activating the ALC and Noise Gate.

SUMMARY

The purpose of this application note is to explain and clarify the setup of the ALC feature provided in Wolfson's CODECs and ADCs.

Recommended setups have been provided as a base to work from. The resultant effect of the ALC is fairly subjective and is likely to vary between applications and against different background noise levels. Some further modifications may be beneficial to optimise the feature for a specific application, but the example settings provided should offer suitable results in many cases.

Specific device datasheets should be referred to for register settings and any additional functions not discussed in this document when setting up the ALC feature.

Setting up the ALC requires an understanding of the type of signal that requires to be accommodated. It also requires an understanding that it is not possible for a single ALC configuration to respond optimally to every type of input. The final implementation is likely to be a compromise, based on an understanding of both the predictable and the unpredictable characteristics of the signal.

APPLICATION SUPPORT

If you require more information or require technical support please contact Wolfson Microelectronics Applications group through the following channels:

Email: apps@wolfsonmicro.com
Telephone Apps: (+44) 131 272 7070
Fax: (+44) 131 272 7001
Mail: Applications at the address on the last page.

Or contact your local Wolfson representative.

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ADDRESS:

Wolfson Microelectronics plc

Westfield House

26 Westfield Road

Edinburgh

EH11 2QB

United Kingdom

Tel :: +44 (0)131 272 7000

Fax :: +44 (0)131 272 7001

Email:: apps@wolfsonmicro.com