

MicroComm DXL

DXL Audio Settings Guide

June, 2021

DXL Audio Settings Guide

Table of Contents

1	INTRODUCTION	1
2	DXL AUDIO PATH	1
3	SETTING DESCRIPTIONS	2
3.1	INPUT (MICROPHONE) SETTINGS	3
3.1.1	Preamp Gain (Levels tab)	3
3.1.2	Microphone (Levels tab)	3
3.1.3	Use AGC (Levels tab)	3
3.1.4	Input Filter (Filters tab)	3
3.2	OUTPUT (SPEAKER) SETTINGS	4
3.2.1	Output Filter (Filters tab)	4
3.2.2	Speaker (Levels tab)	4
3.2.3	Use ANC (Levels tab)	4
3.2.4	Ambient Noise Threshold (Levels tab)	4
4	EXAMPLE SETTINGS	5
4.1	HARDING MASTER STATIONS	5
4.2	HARDING 400-SERIES STATIONS	6
4.3	HARDING 401-SERIES STATIONS	7
4.4	HARDING 300-SERIES 25V STATIONS USING 45 OHM SPEAKER AND CIRCUIT BOARD	8
4.5	HARDING 302-SERIES 25V STATIONS USING 8 OHM SPEAKER AND TRANSFORMER	9
4.6	25V STATIONS WITH LARGE SPEAKER MAGNETS OR CONES (QUAM, DUKANE, ETC.)	10
4.7	25V STATIONS WITH SMALL SPEAKER MAGNETS AND CONES	11
4.8	HARDING 1-GANG INTERCOMS (ICE-41X, ICE-31X)	12
4.9	HARDING 600-SERIES STATIONS	13
4.10	PAGING SPEAKERS ON TBE-310 AND PTA-620	14

1 Introduction

This application note has some basics on the DXL system audio path and examples of station settings for various types of intercom stations.

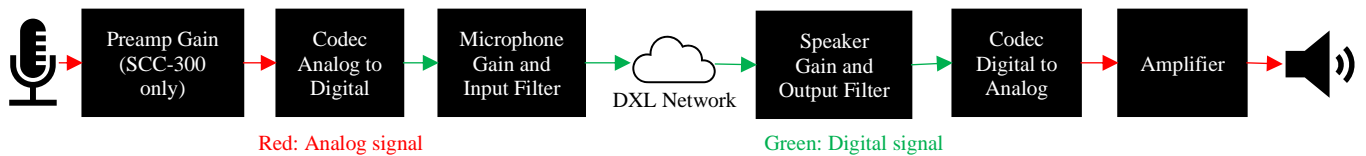
You should have some knowledge of the DXL Administrator Software before using this guide.

2 DXL Audio path

The DXL system processes audio at various steps in the audio transmission process.

First, the microphone device (master when the master is speaking, station when the master is listening) will apply a hardware Preamp Gain if applicable (on stations on 300-series SCC cards). The audio then gets digitized with a codec, then the digital signal processor (DSP) applies a software Microphone gain adjustment and Input digital filter to the digital audio stream, resulting in a digital audio stream input to the system.

The audio stream is transmitted through the system to where the destination speaker device is located. The local DSP at the destination applies a software Speaker gain adjustment and Output digital filter to the stream before sending it to the codec which then converts it to an analog signal. This then gets sent to the output device's amplifier for output to the speaker.



You want to normalize the different devices' input microphone settings to the same levels, so that the input would be similar no matter what device it is (master microphone, 400-series station speaker as microphone, generic 25V station speaker as microphone, etc.). Then with the microphone input being normalized, the output device speaker settings will be set to match the overall loudness and audio characteristics of the specific the room that the speaker is in.

When a device is the sound source or microphone, the "Microphone" gain and "Input" filter settings are used. When a device is the sound destination or speaker, the "Speaker" gain and "Output" filter settings are used.

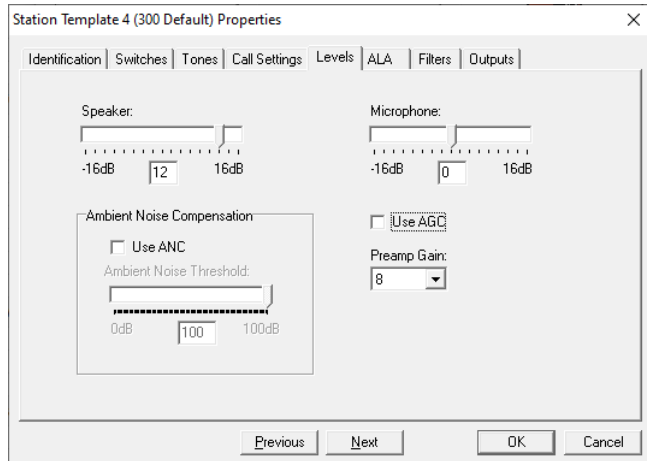
Note that most DXL masters' input settings are already normalized in the master station hardware, so masters generally do not need much adjustment. Stations will vary highly depending on what kind of station it is, the speaker properties and size, backbox type and location, room acoustics, etc., so most of the adjustments you will do will be in the station template (or station) settings.

3 Setting Descriptions

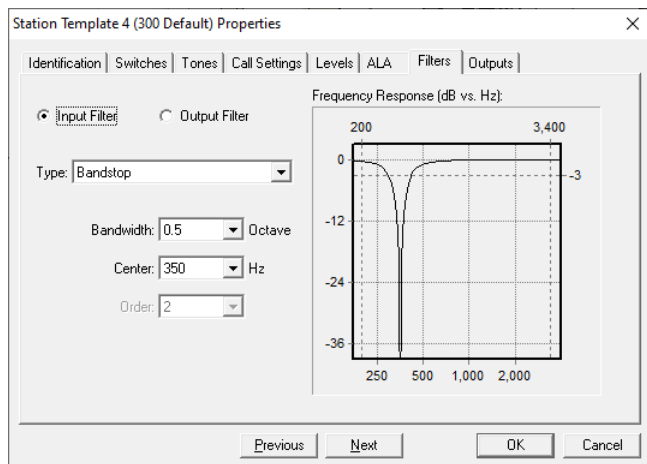
The audio settings are generally divided into two tabs in the configuration.

The Levels tab allows you to specify the input (microphone) and output (speaker) gain settings.

It will also allow you to select Automatic Gain Control (AGC) to adjust the microphone gain level downwards based on average sound power at the microphone when someone is speaking loudly into the microphone, and Ambient Noise Compensation (ANC) to adjust the speaker gain level upwards if the average sound level in that room is loud.



The Filters tab allows you to apply input (microphone) and output (speaker) filters to the device. You can select which filter you are viewing or modifying with the "Input Filter" or "Output Filter" selection. Note that the "Filters" tab is not present on IP master types as the masters are already normalized in the hardware.



3.1 Input (microphone) settings

3.1.1 Preamp Gain (Levels tab)

This is the hardware gain control before digitizing the signal. This is on 300-series station card ports only, as other Harding devices are normalized in the hardware already. You want to set this high enough to get a clear signal without clipping. A setting too low will mean you need more software gain, resulting in quantization errors and a higher noise floor.

3.1.2 Microphone (Levels tab)

This is the software gain control after digitizing the signal. You may want to adjust this based on the microphone position in the room (for example, a speaker in the ceiling, further away from the user than a typical intercom station, may require a higher “Microphone” gain setting).

3.1.3 Use AGC (Levels tab)

This will apply automatic gain control if this is checked. This will cut the microphone level if the average voice level of this device microphone is high, based on a “rolling average” of the microphone level when in a call.

An example of usage would be at a master station that is staffed by different people throughout the day. If a quiet officer staffs this station, the rolling average sound power at the microphone when they are speaking is low, and the system will keep the microphone gain at the “Microphone” setting. If a loud officer staffs the station, the rolling average sound power at the microphone when they are speaking will be higher, and the system will cut the microphone gain. However, after the staff change, it will take some time to adjust to the new average sound power of the staff member. Hence upon staff change after switching from the quiet officer (with the normal microphone gain) to the loud officer, the first few seconds the loud officer speaks will be loud reflecting the sound level of the loud officer. As the system averages the new rolling average microphone sound power level from the loud officer, the system will apply a cut to the microphone level, at least until the next staff change. The reverse may happen when switching from a loud staff member to a quiet staff member, in which case the microphone will start off too low as the volume was cut for the loud staff member, then gradually get back to normal for the quiet staff member.

It is recommended to normally leave this unchecked unless you have a specific use case for this.

3.1.4 Input Filter (Filters tab)

This specifies the filter applied to the microphone signal. You may need to adjust this based on the speaker type. For Harding stations, while the station microphone response is generally optimized to be flat already, there is a resonant frequency of the speaker (which may be noticeable as a “buzzing” sound only at certain voice frequencies). Applying a notch filter at this resonant frequency will reduce this effect. If the resonant frequency of the speaker is not known, a highpass filter is generally recommended to reduce low frequency noise.

3.2 Output (speaker) settings

The audio then passes through the system, and the destination DSP then processes the signal before sending it to the output codec. The following would be applied on the destination speaker device.

3.2.1 Output Filter (Filters tab)

This is the filter applied to the speaker signal. It is generally recommended to apply a highpass filter for most station devices, although for some stations, you may need to apply a different filter such as a bandpass filter, depending on the speaker characteristics. The highpass filter has numerous uses. It reduces low frequency noise, and for speakers with transformers (which are stations on SCC-300 and SCC-400 station cards and TBE talkback expanders), this prevents transformer saturation with inaudible low frequency signals (which can cause power or reset issues if left unfiltered).

Also, because 400-series stations use DC signalling for the switch detection on the one-pair stations, the highpass filter may also reduce “phantom call requests” on 400-series stations particularly when multiple stations are part of a page zone, as otherwise without a filter, a call with low frequency components may be interpreted as a switch action.

3.2.2 Speaker (Levels tab)

This is a software gain control applied to the signal before sending it to the codec to be converted to analog. You will adjust this based on the volume you want at the speaker. Speakers in a small room may need this adjusted lower, while speakers in a large room, or speakers further away from the listener, may need this adjusted higher. +12 dB is a recommended starting point.

3.2.3 Use ANC (Levels tab)

Automatic Noise Compensation (ANC) is used to adjust the speaker volume upwards if the average background sound power in that room is louder than a specified threshold.

If this setting is check marked, ANC will be used.

ANC requires an enhanced process control card (DCC or DCE with “ALA” option) as the system has to constantly monitor the room to determine the average sound level in this case, requiring the enhanced digital signal processing for this function. ANC will not operate on a standard (non ALA) DCC or DCE (this setting will be ignored in this case).

3.2.4 Ambient Noise Threshold (Levels tab)

This is used to set the background noise threshold for the ANC function to take effect, when “Use ANC” is check marked. When the average sound power in the room is higher than this threshold, the ANC will boost the output (speaker) volume for this station by the average sound level exceeding the threshold. For example, if the threshold is set to 70 dB, and the average sound level heard in this room is 75 dB, then the speaker gain will be adjusted by 5 dB (75 dB average - 70 dB threshold)

For more details on ANC operation, you can refer to the Ambient Noise Compensation Functional Specification (FS6800142).

4 Example Settings

The following examples are recommended starting points for audio settings.

Note that master station filter settings are normally already set to reasonable defaults so do not need adjustment unless you need to adjust for specific room acoustics. You will probably want to set the “Speaker” volume to +12 dB for a greater adjustment range

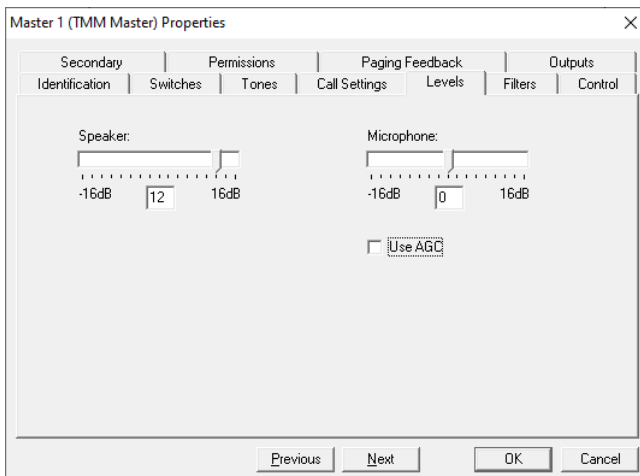
The example settings below are mostly sample settings for “Station templates” (or “Stations” if adjusted individually), since the types and acoustic room properties where stations are located are highly variable, with one example for all types of masters showing the +12 dB “Speaker” volume adjustment.

4.1 Harding Master Stations

Harding Master stations are generally optimized for use with the Harding system.

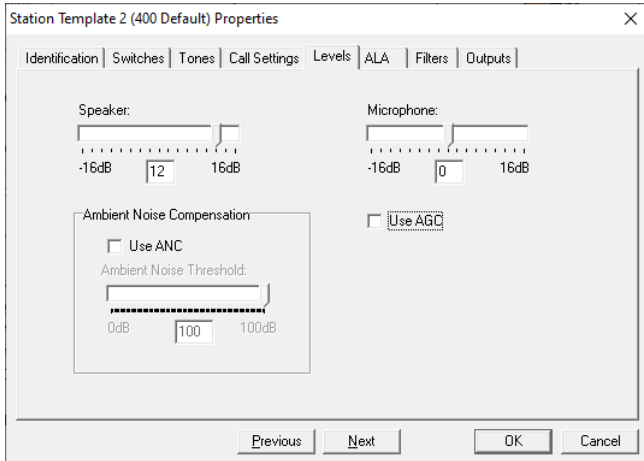
Analog master stations (“TMM Master”, “IMS Master”, “TSM Master”) have filter settings on the “Filters” tab, but you can generally leave these set to no filter (Input Filter and Output Filter can be left as “None”) unless you need to adjust for room acoustics.

VoIP master stations (“IP Master”, and “TMM IP Master”) do not have a “Filters” tab. There are some adjustments on the web configuration interface for these master types.

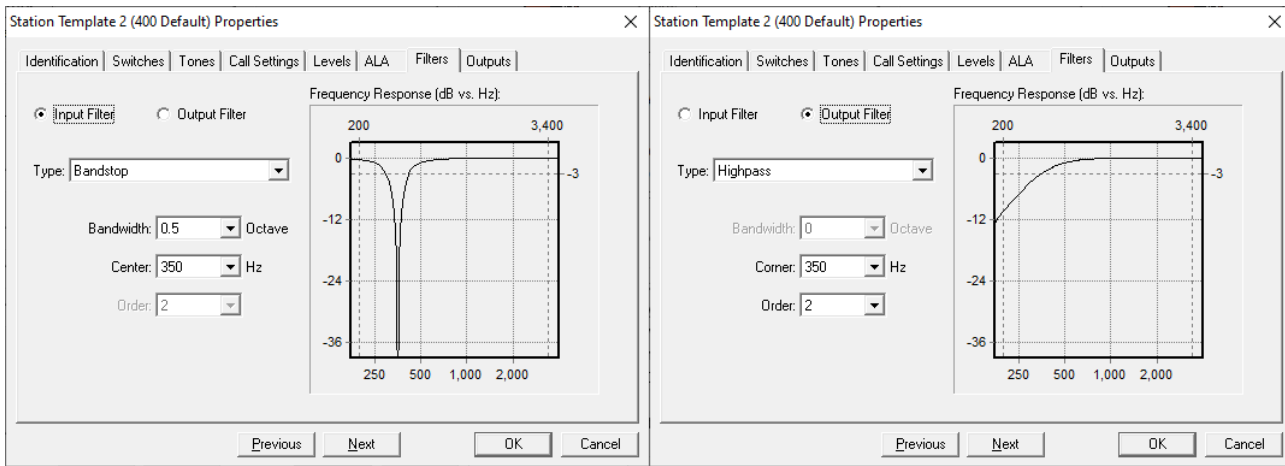


You would generally only need to set the Levels tab to adjust the speaker volume for the requirement for the room. A starting point of +12 dB is recommended, which allows for a wide range of volume adjustment with the master’s volume control.

4.2 Harding 400-series stations



The Levels tab would generally adjust the speaker volume for the requirement for the room. A starting point of +12 dB is recommended, which makes the default speaker level approximately the same as if there was a person in the room standing the same distance away as the speaker.

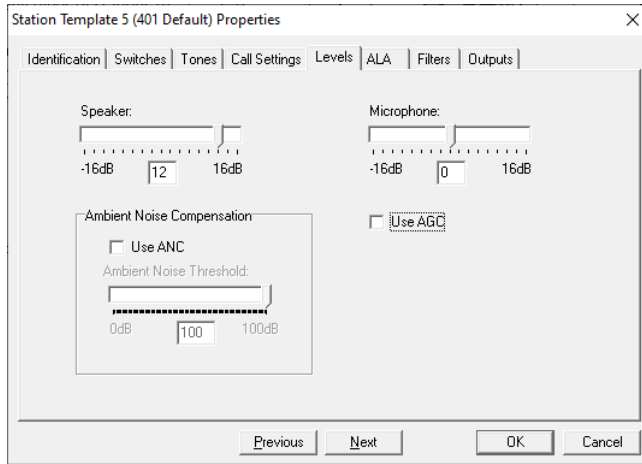


While the 400-series stations are optimized for a flat response, it is recommended to put a 1/2 octave bandstop filter at 350 Hz as the input filter. This would reduce the signal at the 350 Hz resonant frequency of the speaker (which may cause a “buzzing” noise at certain frequencies if no filter was applied).

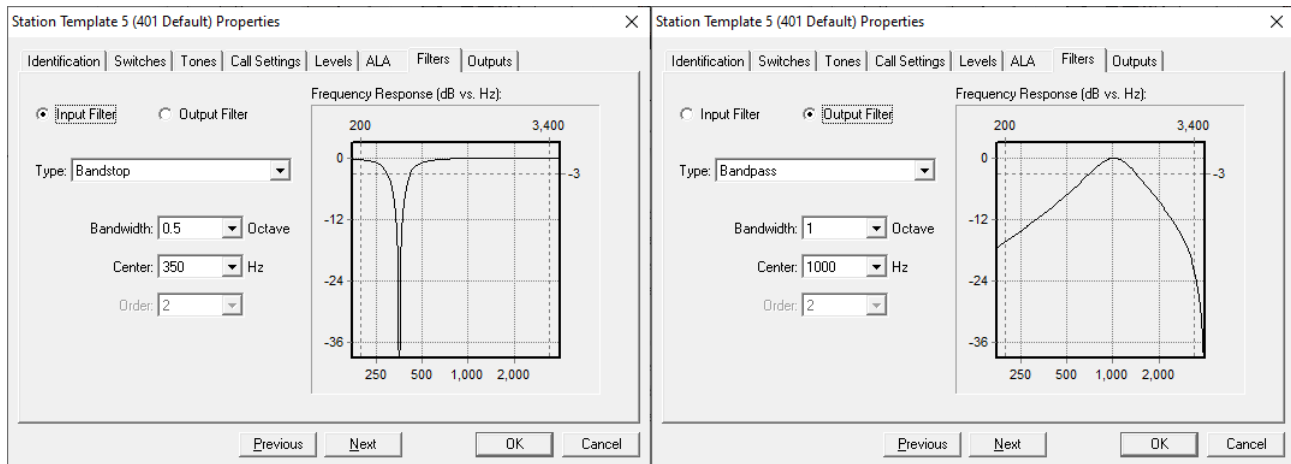
It is recommended to put a 2nd order highpass filter at 350 Hz corner frequency as the output filter to filter out low frequency noise and to prevent transformer saturation from inaudible low frequencies (which if unfiltered may cause power or reset issues on the DCC or DCE).

4.3 Harding 401-series stations

Harding 401-series stations have an LED which is driven by low frequency FSK over the audio pair. The audio characteristics are otherwise similar to 400 series stations with the exception of the Output filter.



The Levels tab would generally adjust the speaker volume for the requirement for the room. A starting point of +12 dB is recommended, which makes the default speaker level approximately the same as if there was a person in the room standing the same distance away as the speaker.



While the 401-series stations are optimized for a flat response, it is recommended to put a 1/2 octave bandstop filter at 350 Hz as the input filter. This would reduce the signal at the 350 Hz resonant frequency of the speaker (which may cause a “buzzing” noise at certain frequencies if no filter was applied).

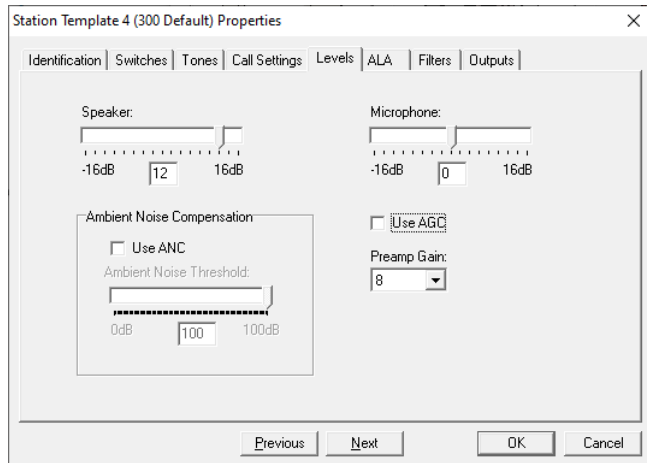
Because the 401-series LED and direction control use a low frequency FSK signal, the 401-series station electronics are more sensitive to lower frequencies and any extraneous signals need to be filtered out or the LED may be erratic and the output audio may cut in and out. It is recommended to put a 1 octave bandpass filter centered at 1000 Hz as the output filter to filter out any audio frequencies that may interfere with the direction control signaling and LED operation.

DXL Audio Settings Guide

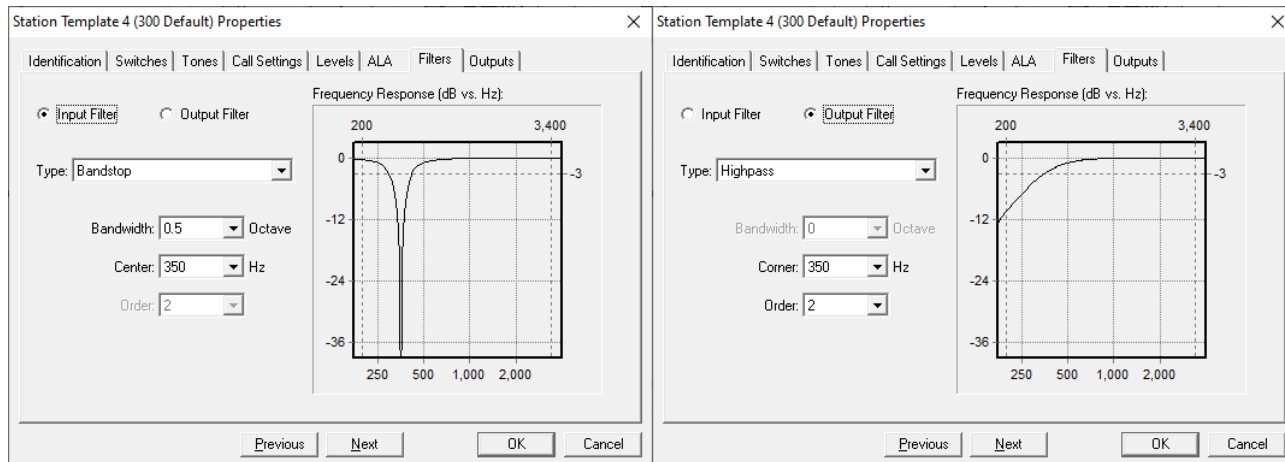
4.4 Harding 300-series 25V stations using 45 ohm speaker and circuit board

The standard Harding 300-series stations have a 45 ohm speaker, and a transformer mounted on a small circuit board. The ordering part number would start with ICE-3x0 or ICM-3x0 and the part number on the faceplate would start with ASM66.

Stations on SCC-300 station cards have a preamp gain option to accommodate for hardware microphone gain depending on the speaker characteristics. Harding 300-series stations would have a preamp gain set to 8 for this speaker's characteristics.



The other settings on the Levels tab would generally adjust the speaker volume for the requirement for the room. A starting point of +12 dB is recommended, which makes the default speaker level approximately the same as if there was a person in the room standing the same distance away as the speaker.



While the 300-series stations are optimized for a flat response, it is recommended to put a $\frac{1}{2}$ octave bandstop filter at 350 Hz as the input filter. This would reduce the signal at the 350 Hz resonant frequency of the 45 ohm speaker (which may cause a “buzzing” noise at certain frequencies if no filter was applied).

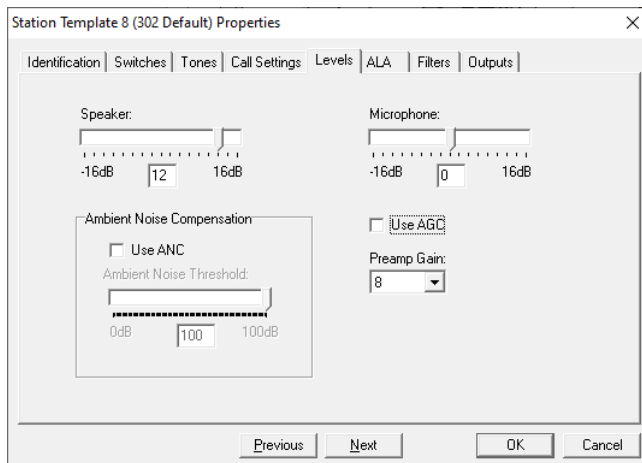
It is recommended to put a 2nd order highpass filter at 350 Hz corner frequency as the output filter to filter out low frequency noise and to prevent transformer saturation from inaudible low frequencies (which if unfiltered may cause power or reset issues on the DCC or DCE).

DXL Audio Settings Guide

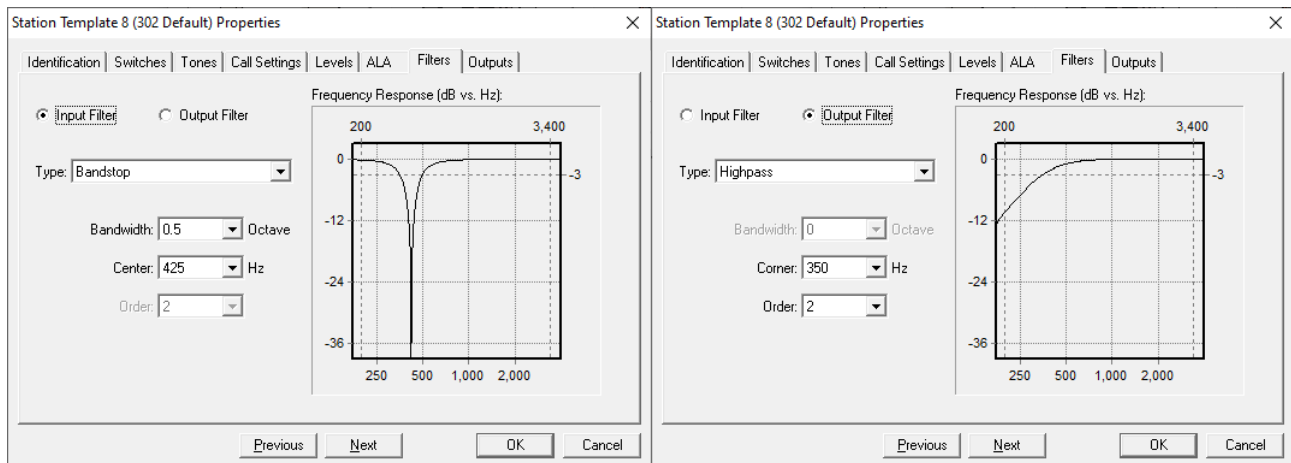
4.5 Harding 302-series 25V stations using 8 ohm speaker and transformer

Harding 302-series stations have an 8 ohm speaker and a larger transformer, with no circuit board. The ordering part number would start with ICE-3x2 and the part number on the faceplate would start with ASM2059.

Stations on SCC-300 station cards have a preamp gain option to accommodate for hardware microphone gain depending on the speaker characteristics. Harding 302-series stations would have a preamp gain set to 8 for this speaker's characteristics.



The other settings on the Levels tab would generally adjust the speaker volume for the requirement for the room. A starting point of +12 dB is recommended, which makes the default speaker level approximately the same as if there was a person in the room standing the same distance away as the speaker.



While the 302-series stations are optimized for a flat response, it is recommended to put a $\frac{1}{2}$ octave bandstop filter at 425 Hz as the input filter. This would reduce the signal at the 420 Hz resonant frequency of the 8 ohm speaker (which may cause a “buzzing” noise at certain frequencies if no filter was applied).

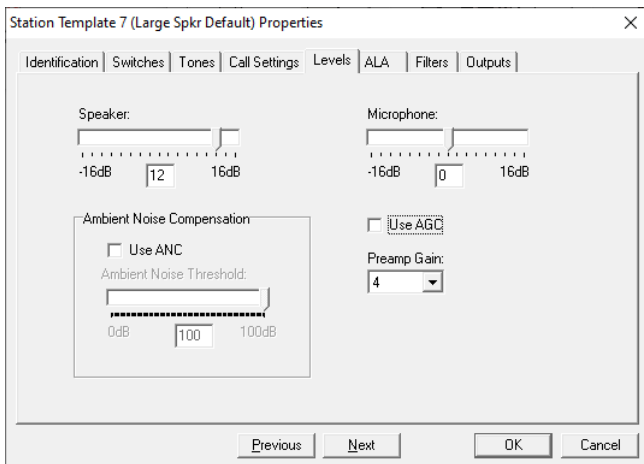
It is recommended to put a 2nd order highpass filter at 350 Hz corner frequency as the output filter to filter out low frequency noise and to prevent transformer saturation from inaudible low frequencies (which if unfiltered may cause power or reset issues on the DCC or DCE).

DXL Audio Settings Guide

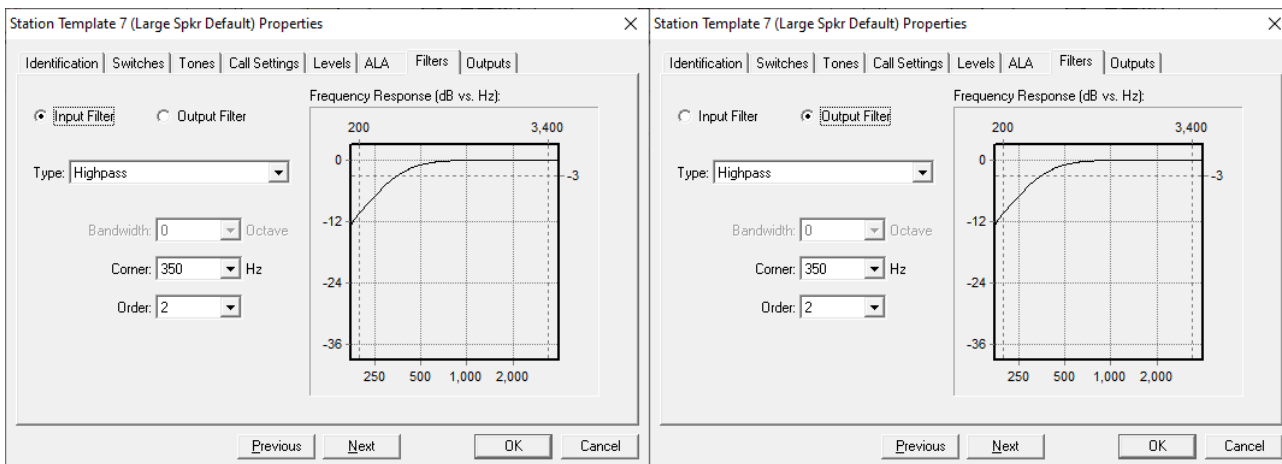
4.6 25V stations with large speaker magnets or cones (Quam, Dukane, etc.)

Many generic 25V intercom stations from other manufacturers have a larger size speaker magnet (typical for Quam and Dukane stations), or have a larger speaker cone speaker greater than 4 inches. These speakers can generate a stronger microphone signal.

Stations on SCC-300 station cards have a preamp gain option to accommodate for hardware microphone gain depending on the speaker characteristics. You can start with a lower preamp gain of 4. We recommend the speakers be tapped at 1 watt, and if you require less volume, you can adjust the Speaker volume appropriately. If you tap the speaker at a lower power, then you will not be able to adjust the volume much higher with software, while at a 1 watt tap you can always adjust the volume lower in software.



The other settings on the Levels tab would generally adjust the speaker volume for the requirement for the room. A starting point of +12 dB is recommended.



If you know the resonant frequency of the speaker, you could set a bandstop input filter similar to a 300 series station, but if you do not know this, you can start with a 2nd order highpass filter with 350 Hz corner frequency.

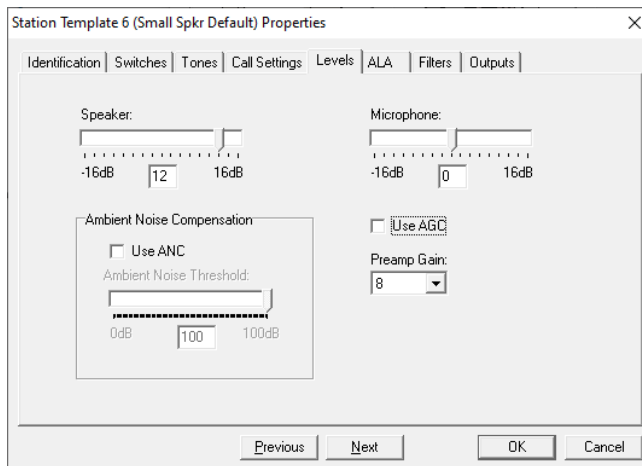
It is recommended to put a 2nd order highpass filter at 350 Hz corner frequency as the output filter to filter out low frequency noise and to prevent transformer saturation from inaudible low frequencies (which if unfiltered may cause power or reset issues on the DCC or DCE).

DXL Audio Settings Guide

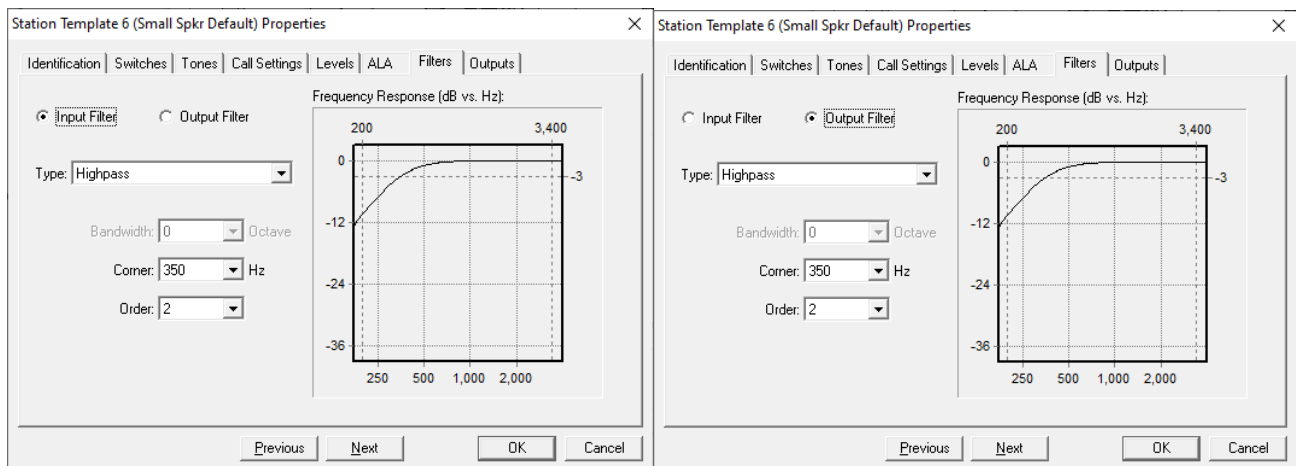
4.7 25V stations with small speaker magnets and cones

Some less common 25V intercom station types have a smaller speaker magnet and hence are less sensitive as a microphone.

Stations on SCC-300 station cards have a preamp gain option to accommodate for hardware microphone gain depending on the speaker characteristics. You may want to start off by setting the preamp gain to 8 to account for the low sensitivity of the speaker as a microphone. We recommend the speakers be tapped at 1 watt, and if you require less volume, you can adjust the Speaker volume appropriately. If you tap the speaker at a lower power, then you will not be able to adjust the volume much higher with software, while at a 1 watt tap you can always adjust the volume lower in software.



The other settings on the Levels tab would generally adjust the speaker volume for the requirement for the room. A starting point of +12 dB is recommended.



If you know the resonant frequency of the speaker, you could set a band stop input filter similar to a Harding 300-series station, but if you do not know this, you can start with a 2nd order highpass filter with 350 Hz corner frequency.

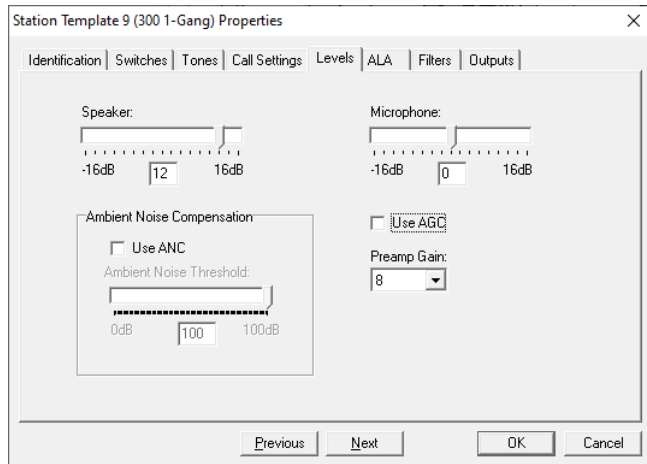
It is recommended to put a 2nd order high pass filter at 350 Hz corner frequency as the output filter to filter out low frequency noise and to prevent transformer saturation from inaudible low frequencies (which if unfiltered may cause power or reset issues on the DCC or DCE).

DXL Audio Settings Guide

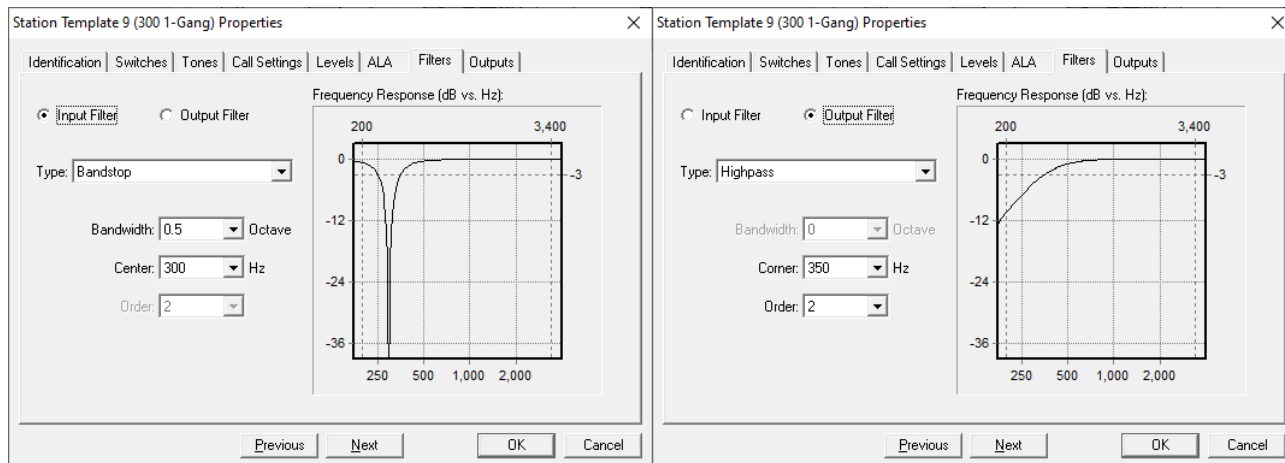
4.8 Harding 1-Gang Intercoms (ICE-41x, ICE-31x)

Harding 1-gang intercoms have a small speaker and transformer in order to fit a 1-gang backbox.

For a 1-gang intercoms on 300-series SCC cards, there is a preamp gain option to accommodate for hardware microphone gain depending on the speaker characteristics. Harding 300-series 1-gang stations would have a preamp gain set to 8 for this speaker's characteristics. A 400-series 1-gang intercom would not have the Preamp Gain option shown.



The other settings on the Levels tab would generally adjust the speaker volume for the requirement for the room. A starting point of +12 dB is recommended, which makes the default speaker level approximately the same as if there was a person in the room standing the same distance away as the speaker.

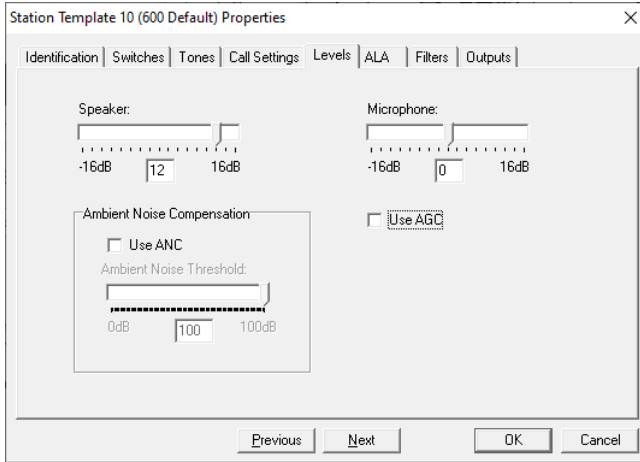


While the 1-gang stations are optimized for a flat response, it is recommended to put a 1/2 octave bandstop filter at 300 Hz as the input filter. This would reduce the signal at the 260 Hz resonant frequency of the 1-gang speaker (which may cause a “buzzing” noise at certain frequencies if no filter was applied).

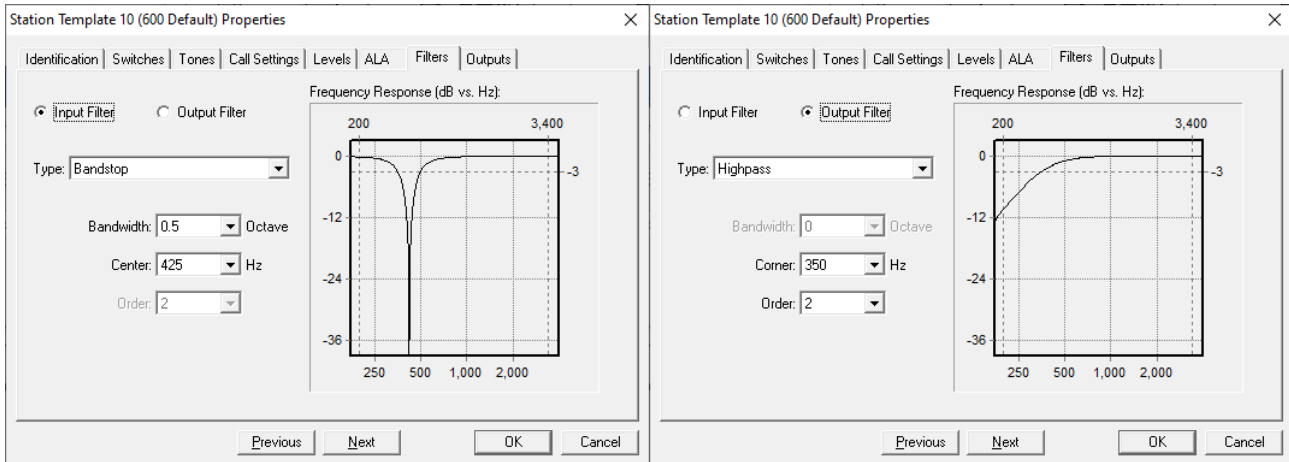
It is recommended to put a 2nd order highpass filter at 350 Hz corner frequency as the output filter to filter out low frequency noise and to prevent transformer saturation from inaudible low frequencies (which if unfiltered may cause power or reset issues on the DCC or DCE).

4.9 Harding 600-series stations

Harding 600 series stations are IP stations that use an 8 ohm speaker so are similar to the 302-series stations in audio properties.



The Levels tab would generally adjust the speaker volume for the requirement for the room. A starting point of +12 dB is recommended, which makes the default speaker level approximately the same as if there was a person in the room standing the same distance away as the speaker.

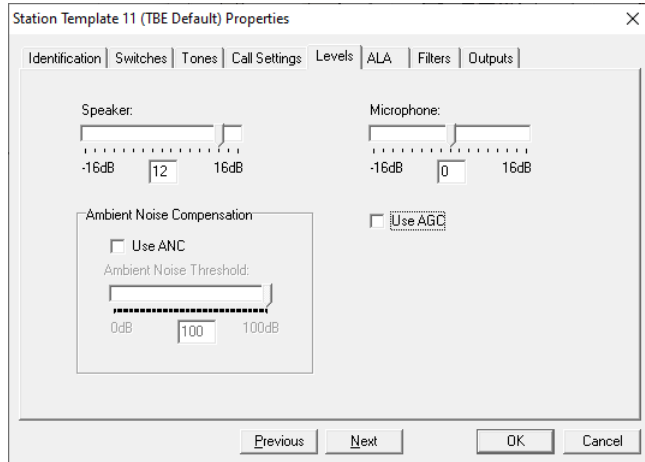


While the 600-series stations are optimized for a flat response, it is recommended to put a 1/2 octave bandstop filter at 425 Hz as the input filter. This would reduce the signal at the 420 Hz resonant frequency of the 8 ohm speaker (which may cause a “buzzing” noise at certain frequencies if no filter was applied).

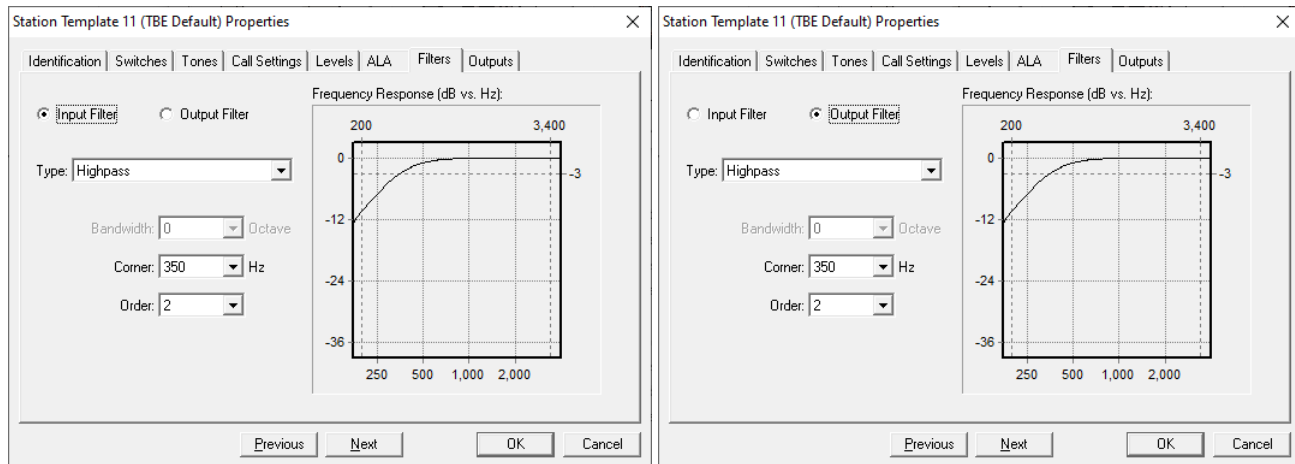
You can optionally put a 2nd order highpass filter at 350 Hz corner frequency as the output filter to filter out low frequency noise. Since these speakers do not use a transformer, this is not necessary to prevent transformer saturation unlike the analog intercom types.

4.10 Paging speakers on TBE-310 and PTA-620

Paging speakers are probably the most varied audio component and the settings here are only starting recommendations. Paging speakers are of varied types (cone speakers, horn or folded horn speakers, soundsphere speakers, etc.), may be in various orientations and distance from the listeners, are of varied power, and may have multiple speakers chained together on one circuit.



It is recommended to have a +12 dB Speaker gain starting point, which you can adjust up or down depending on the sound level you need for that paging circuit. When you are using a TBE or PTA speaker for talkback, you may need to make adjustments to the Microphone setting to reflect the microphone sensitivity, room location, etc.



The filter settings will also vary depending on the type of speaker you have. Like 300-series stations, if you know the resonant frequency of the speaker, you could add a bandstop input filter at the resonant frequency, but if you do not know this, a good starting point is to set a 350 Hz highpass filter to begin with.

It is recommended to start with a 2nd order highpass filter at 350 Hz corner frequency as the output filter to filter out low frequency noise and to prevent transformer saturation from inaudible low frequencies (which if unfiltered may cause power or reset issues on speakers on a TBE). You may need to adjust the filter settings depending on the speaker characteristics.