

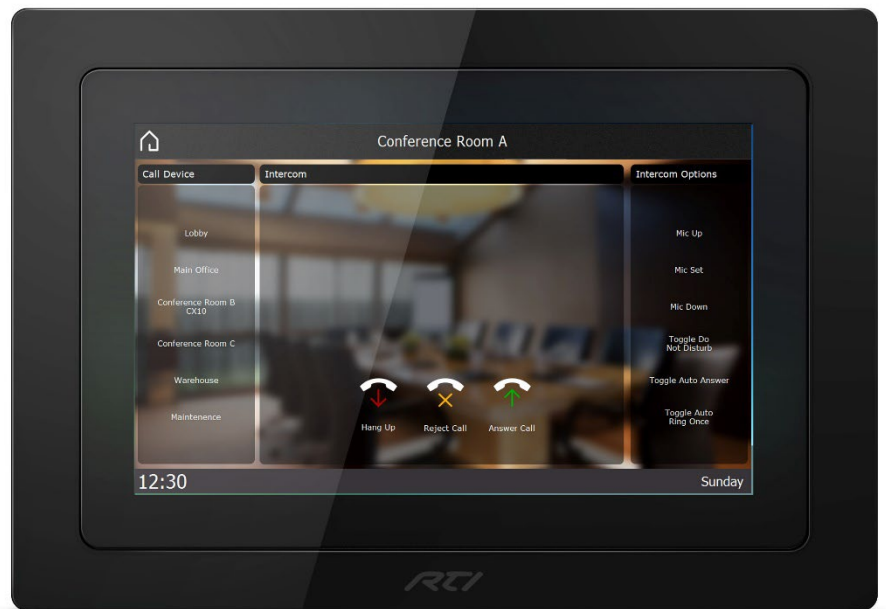
Intercom Setup & Calibration Tips

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A. Overview

The codecs supported for VOIP / Intercom Audio are G.711-A-law audio & G.711-u-law audio; it is recommended to verify that the 3rd party SIP intercom is utilizing the codecs if it is available as a setting option. If any internal settings changes cannot improve the audio and a third-party SIP product is not utilized, it is best to try setting up the audio from a similar device. For example, you may use a KX4 communicating with a KX10s or a KX10s to KX10s. Retest again with a third-party SIP and see if it has the same results in quality. This process will help determine what device requires adjustment and possibly has hardware issues. Testing with the pre-programmed templates usually works best in this scenario, communication between RTI only devices allows the best result before implementing any 3rd party SIP.

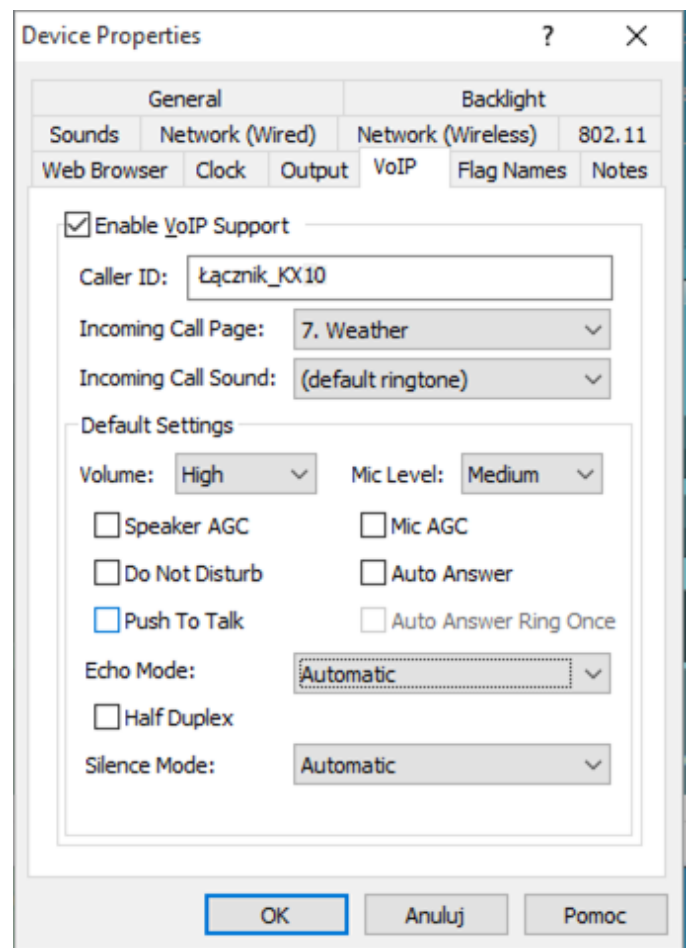


B. Settings

As far as the standard settings adjustments go, these are the basic tips: It may be possible that the Mic/Volume output needs to be adjusted for threshold, response, etc., based on where the units are located, how far away the user will be speaking to them, and background noise. In a proper test environment, optimal performance was possible with all the intercom-capable devices, so it may just require some fine-tuning to help match the environment. The location has thus far played a significant role in the quality of the audio picked up at the receiver. Some cases may require setup in a controlled environment and only minor adjustments made at the installation location.

- The default settings are recommended within the VOIP properties, but you may need to adjust the audio/threshold/response, etc., live to improve performance.

For adjusting the microphone & the audio quality settings for response time, volume, etc. can all be found within the Intercom variables.



C. Testing & Adjusting

The programmer should create a page like the example below for ease of calibration if you would like to create an active deployment. Please keep in mind that it should not be necessary to adjust all the optional tools, creating more disparity in sound integrity.

In testing environments, it has been noted that most issues are related to audio feedback from local sources and only require that the threshold levels may need adjustment following the applicable Mic volume / Speaker.

Testing this from RTI Device to RTI device (same preferable models) will help find the best levels, so only specific settings require adjustment with other products.

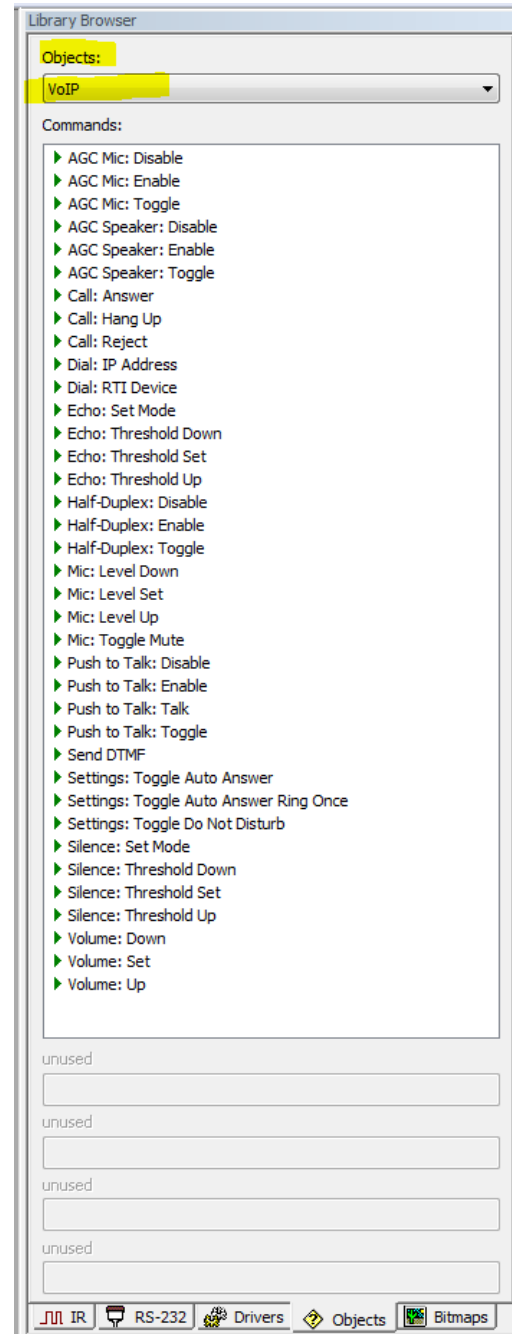
The screenshot displays the 'Intercom Settings' interface. At the top, it shows 'Call Duration: 10' and 'Caller: 192.168.1.100'. The interface is divided into several sections:

- Call Settings:** Includes buttons for 'Call Settings', 'Answer', 'Hang Up', and 'Reject'. It also features a 'Call Settings' menu and several status indicators (Disabled, Fixed).
- SETTINGS:** Contains buttons for 'AGC Mic', 'AGC Speaker', 'Echo - Set Mode', 'Half-Duplex', and 'Silence Mode'. It also has 'Tx Frame Size' and 'Tx Frame Rate' options with values like 128 x 96, 176 x 144, 320 x 240, and 352 x 288.
- VIDEO:** Includes 'Auto Send Off', 'Rx Video Disabled', 'Tx Video Disabled', 'Auto TX Video', 'RX Video', and 'RX Video' buttons.
- FEEDBACK:** A table of status indicators for various settings like 'Auto Answer Off', 'Ring Once Off', 'DND Off', 'Half-Duplex Disabled', 'Mute Off', 'AGC Mic Not Enabled', '75 - Sil Suppress Threshold', '50 - Tx Audio Lev', '320 - Rx Vid Act Frame Width', '240 - Rx Video Act Frame Height', '320 - Tx Video Actual Frame Width', '240 - Tx Video Actual Frame Height', '15 - Tx Video Actual Frame Rate', '192.168.1.100 - Caller', '10 Duration', '50 - Mic Level', '50 - Vol', '75 - Echo Suppress Threshold', 'Fixed - Echo Suppress Mode', 'Echo Suppress Not Enabled', '50 - Rx Audio Lev', 'false - Rx Vid Enabled', 'Tx Video Disabled', '15 - Rx Vid Act Frame Rate', '320 Tx Vid Max Frame Width', '240 - Tx Video Max Frame Height', and '15 - Tx Video Max Frame Rate'.
- Other Elements:** Includes 'Incoming Video' window, 'Push To Talk' and 'Talk' buttons, and a 'FEEDBACK' section with a speaker icon.

If it needs to be turned down, the microphone threshold can be adjusted within the VOIP settings. There are many different audio/mic settings you can adjust for this, the one that usually helps with reducing background noise is usually in the “Echo” settings:



Other specific level settings can be added manually from the VOIP objects library:



D. Notes

Note 1:

It is also a good measure to attempt testing the units at a reasonable distance from each other to avoid possible feedback that may cause the Echo suppression/delay to be overreactive.

In the case of audio cutting out or unusual sporadic non-response, it is recommended to confirm that there is no Network/ Lan or ethernet related equipment, configurations, or transmission failure limiting the ability of the VOIP to send/ receive optimally between IP devices.

If only using RTI devices, the trouble may be device-specific. In most cases witnessed, this is usually background noise (location), power-related, firmware, or a hardware issue with the speaker or connections. It becomes difficult to solve when attempting too many changes to the audio and then to improve the quality from that point. Reloading firmware will reset the values. Next, they will send the configuration file with the VOIP settings to “push to talk” (disable AGC).

This process is just a direct device-to-device connection with a standard voice. Installers can verify the connection and sound and then make changes if necessary. For most situations, we recommend setting the properties to half-duplex with AGC not on (amplifies the sound), as it has had favorable results. This setting serves as a reasonable starting point, and if there’s a speaker issue somewhere or a network connectivity problem, issues affecting the sound can be found easier.

Note 2:

We recommend a third-party SIP server using more than a point-to-point system (two RTI SIP devices). KA to KA will not work well with this generation controller due to issues with noise cancellation.

Last, please ensure you understand the codecs that are supported when working with third-party products. Here is a list of the audio and video codecs that are supported.

Audio codec supported:

Speex PCMU PCMA Opus

Video codec support:

H.264 H.265 VP8

Contact the manufacturer if you are uncertain about the codecs supported by your device.