Grandstream Networks, Inc.

GXW42xx/HT8xx Alarm System

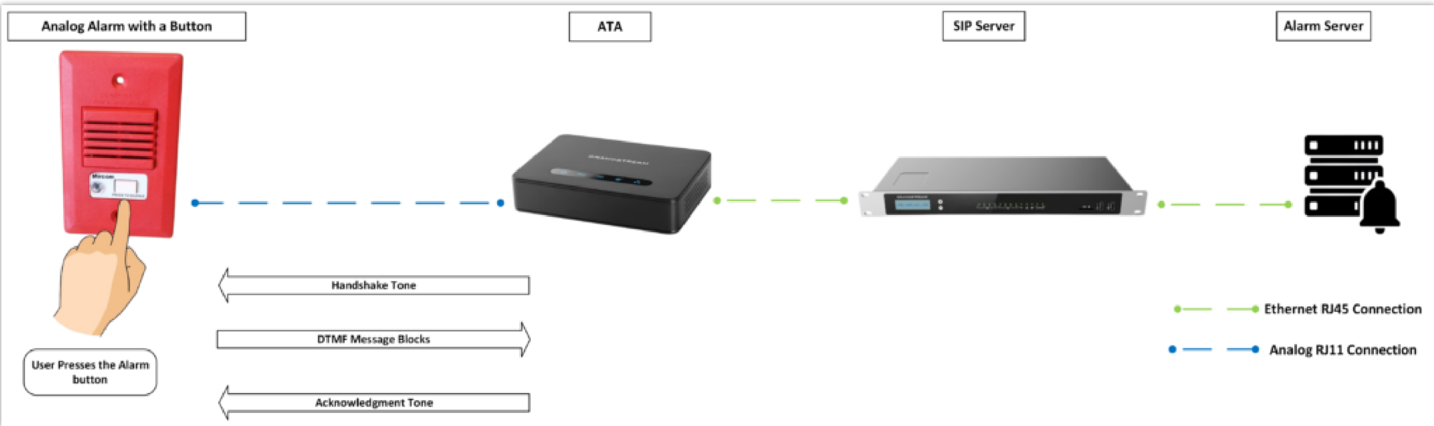
Configuration Guide

GXW42xx/HT8xx Alarm System Guide

Introduction

This guide provides an overview for configuring a VoIP gateway or ATA for an alarm system with different DTMF tone transmission modes. The settings discussed are done to optimize performance and resolve common issues related to alarm system communication. The focus is on configuring devices for compatibility with RFC2833, InAudio, and SIP INFO modes.

Below is an example diagram that illustrates a common use case scenario where an analog phone will be used with a GXW42xx/HT8xx device, please note that this is just an example and each setup can be customized based on the deployment's needs:



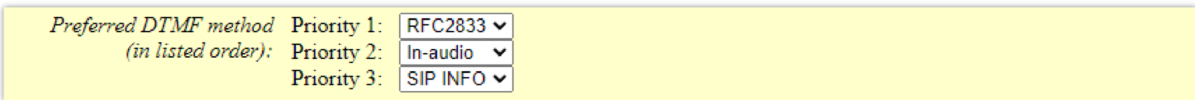
Alarm System Diagram

- **Handshake Tone:** This is the initial signal sent between the Analog Alarm Device and the Alarm Server to establish communication. It ensures both ends are ready for data transfer, similar to a "hello" that confirms the devices are connected.
- **DTMF Message Blocks:** After the handshake, the Alarm Device sends the actual information (alarm data) in the form of DTMF (Dual-Tone Multi-Frequency) tones, the method of transmitting the DTMF can differ, we will discover each method separately in this guide.
- **Acknowledgment Tone:** Also called "Kissoff Tone", once the Alarm Server successfully receives and processes the DTMF message blocks, it sends back a "Kissoff tone" to the Alarm Device to acknowledge that the message was received correctly.

How It Works

Alarm systems or Analog phones use VoIP gateways to transmit signals. These systems rely on DTMF (Dual-Tone Multi-Frequency) tones to relay commands or acknowledgment codes. For the HT8xx ATAs and GXW42xx gateways, DTMF tones can be transmitted using RFC2833, InAudio, or SIPINFO modes.

In RFC2833 mode, the tones are sent as part of the RTP (Real-Time Protocol) stream, which allows for better handling in VoIP systems. In InAudio mode, the DTMF tones are transmitted within the audio stream, this will require special settings for pass-through. SIP INFO mode uses SIP (Session Initiation Protocol) messages to send the tones out-of-band.



Preferred DTMF method

Configuring DTMF Tones

RFC2833 Mode

The RFC2833 mode is one of the most commonly used methods for transmitting DTMF tones in VoIP environments. Below are the detailed steps and explanations for each configuration field required for RFC2833 mode:

Under **FXS Port → Codec Settings** define the following:

1. RFC2833 Event Count = 3

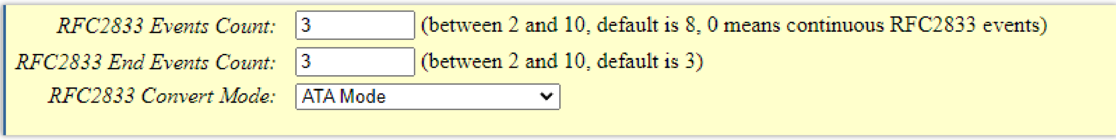
- **What It Does:** This defines how many DTMF packets (or events) are transmitted for each DTMF tone pressed. The number "3" means that three separate packets will be sent per keypress.
- **Why It Matters:** Sending multiple packets increases the reliability of tone transmission. Even if some packets are lost during transmission, the system will still have enough data to recognize the DTMF tone correctly.

2. RFC2833 End Event Count = 3

- **What It Does:** specifies how many packets are sent to indicate the end of the DTMF tone. Like the Event Count, three packets are sent to signal the end of the tone.
- **Why It Matters:** Properly signaling the end of a tone is important to ensure that the receiving system doesn't misinterpret the tone's length or continuation, which enhances accuracy in tone recognition.

3. RFC2833 Convert Mode = ATA Mode

- **What It Does:** This field determines how the gateway processes and transmits DTMF signals. Setting it to ATA (Analog Telephone Adapter) mode means the DTMF tones are converted and handled in a way that's optimized for ATA devices, which are commonly used in alarm systems.
- **Why It Matters:** Using ATA mode ensures compatibility with most alarm systems that rely on analog signal conversion to digital VoIP. It makes communication smoother, especially when interfacing with different networks or providers.



RFC2833 Mode Configuration

ıAudio Mode

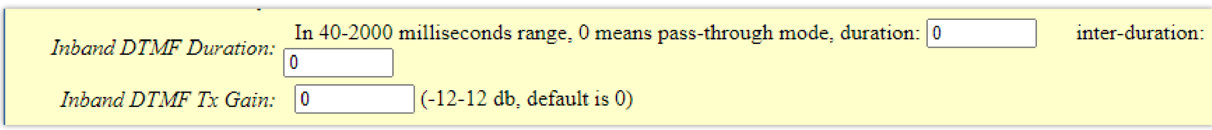
When using **InAudio mode**, DTMF tones are embedded directly into the audio stream. This mode is less commonly used than RFC2833 but may be preferred in certain VoIP configurations. Below are the steps and explanations for configuring DTMF tones in **InAudio mode**:

1. Inband DTMF Duration = 0

- **What It Does:** Setting the "Inband DTMF Duration" to **0** enables **pass-through mode**, meaning that the tones are sent without modification directly through the audio stream. In this mode, the system does not process or alter the tones; they are transmitted exactly as they are received.
- **Why It Matters:** Pass-through mode is critical in **InAudio mode** because it ensures that DTMF tones are treated as regular audio signals, allowing them to be transmitted more accurately. Without this setting, the tones could be distorted or incorrectly interpreted, leading to communication issues with the alarm system.

2. Tone Clarity (*Inband DTMF Tx Gain*) = 0

- **What It Does:** The clarity of the tones in InAudio mode depends on the audio stream quality, configuring the option "inband DTMF Tx Gain" directly impacts tone clarity. If the **Tx Gain** is set too low, the tones may be faint, making them harder for the receiving device to detect. If the gain is too high, it can distort the tones,
- **Why It Matters:** In InAudio mode, DTMF tones are more susceptible to distortion because they are part of the audio signal. Poor network conditions or low-quality audio codecs can cause tones to be misinterpreted. You need to choose audio codecs that maintain the integrity of the audio stream, such as G.711,



Inband DTMF Duration and Tone Clarity

SIP INFO Mode

SIPINFO mode transmits DTMF tones using SIP (Session Initiation Protocol) messages, rather than embedding them in the audio stream. This method is useful when a system cannot reliably handle DTMF tones through audio channels, and it sends the tones out-of-band as control signals. Below are the detailed steps and explanations for configuring DTMF tones in **SIP INFO mode**:

1. SIP INFO Mode Activation

- **What It Does:** SIP INFO mode must be enabled on your VoIP gateway or ATA (Analog Telephone Adapter). This is found in the device's configuration settings under the DTMF transmission options, where you can select SIPINFO as the method for DTMF tone transmission.
- Why It Matters: Enabling SIP INFO mode ensures that DTMF tones are sent as separate SIP messages rather than being embedded in the audio stream or sent via RTP. This method avoids the issues that might arise from poor audio quality or compressed codecs, making it more reliable in networks with variable performance.

2. Configure DTMF Payload Type

- **What It Does:** Some devices allow you to set the payload type for DTMF tones. For SIP INFO mode, the DTMF tones are sent using INFO messages within the SIP signaling, and this typically doesn't require the same payload configuration as in RFC2833 or InAudio modes.
- Why It Matters: While you may not need to adjust the payload type for SIP INFO mode, ensuring your VoIP device is configured to correctly interpret and handle SIP INFO messages is crucial. If your device allows for advanced configuration, consult the user manual to see if there are options related to SIP INFO message handling.

3. Codec Selection (Optional)

- **What It Does:** Although **SIP INFO mode** sends DTMF tones out-of-band, selecting the right codec for voice calls is still important. Use high-quality codecs like **G.711** for clear audio during calls.
- **Why It Matters:** While SIP INFO mode is independent of the audio stream, it's still a good practice to ensure that the audio codec is suitable for maintaining the overall quality of the call. If the VoIP call's voice quality is poor, it might indicate broader network issues that could impact SIP signaling and thus DTMF transmission.

SIP INFO Mode

Supported Devices

Device Name	Firmware Required
HT81x (HT812, HT814, HT818)	1.0.51.1+
HT80x (HT801, HT802)	1.0.51.1+
HT813	1.0.19.1+
GXW42xx V2 (GXW4216 V2, GXW4224 V2, GXW4232 V2, GXW4248 V2)	1.0.23.2+

List of Supported Devices

Need Support?

Can’t find the answer you’re looking for? Don’t worry we’re here to help!

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