



## Usando o GXV3500 para receber ligações de um HT-503 com um viva-voz

Esse tutorial tem como objetivo utilizar a saída de áudio do GXV3500 para a transmissão da voz proveniente de uma linha/ramal do HT503.

### *Configuração do HT-503:*

#### Aba **FXO PORT**

1. Campo ***“Primary SIP server”***: IP do GXV 3500;
2. Campo ***“SIP registration”***: NO
3. Campo ***“Preferred Codec”***: PCMU
4. Campo ***“Number of Rings”***: 1
5. Campo ***“Wait for Dial-Tone”***: NO
6. Campo ***“Stage Method”***: 2

### Grandstream Device Configuration

**STATUS** **BASIC SETTINGS** **ADVANCED SETTINGS** **SFXS PORT** **FXO PORT**

**Account Active:**  No  Yes

**Primary SIP Server:**  (e.g., sip.mycompany.com, or IP address)

**Failover SIP Server:**  (Optional, used when primary server no response)

**Prefer Primary SIP Server:**  No  Yes (yes - will register to Primary Server if Failover registration expires)

**Outbound Proxy:**  (e.g., proxy.myprovider.com, or IP address, if any)

**SIP Transport:**  UDP  TCP  TLS (default is UDP)

**NAT Traversal (STUN):**  No  No, but send keep-alive  Yes

**SIP User ID:**  (the user part of an SIP address)

**Authenticate ID:**  (can be identical to or different from SIP User ID)

**Authenticate Password:**  (purposely not displayed for security protection)

**Name:**  (optional, e.g., John Doe)

**DNS Mode:**  A Record  SRV  NAPTR/SRV

**Tel URI:**

**SIP Registration:**  No  Yes

**Unregister On Reboot:**  No  Yes

**Outgoing Call without Registration:**  No  Yes

**Number of Rings:**  (1-50. Default 4)  
 (Number of rings for a PSTN incoming call before FXO port answers to accept VoIP number)

**PSTN Ring Thru FXS:**  No  Yes (Default Yes)  
 (If set to yes, all incoming PSTN calls will ring the FXS port after the Ring Thru Delay)

**PSTN Ring Thru Delay (sec):**  (1-10 seconds. Default 4 seconds)

**Channel Dialing**

**DTMF Digit Length (ms):**  (40-127 milliseconds, Default 100 milliseconds)

**DTMF Dial Pause (ms):**  (40-127 milliseconds, Default 100 milliseconds)

**First Digit Timeout (sec):**  (1-20 seconds. Default 10 seconds)

**Inter-Digit Timeout (sec):**  (1-15 seconds. Default 4 seconds)

**Wait for Dial-Tone:**  No  Yes (Default Yes - dial upon dial-tone)

**Stage Method (1/2):**  (Default 2 - 2 stage dialing)

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### Aba **BASIC SETTING**S

1. Campo **“Unconditional Call forwarding to VoIP”**: qualquer número
2. Campo **“Sip Server”**: IP do GXV3500

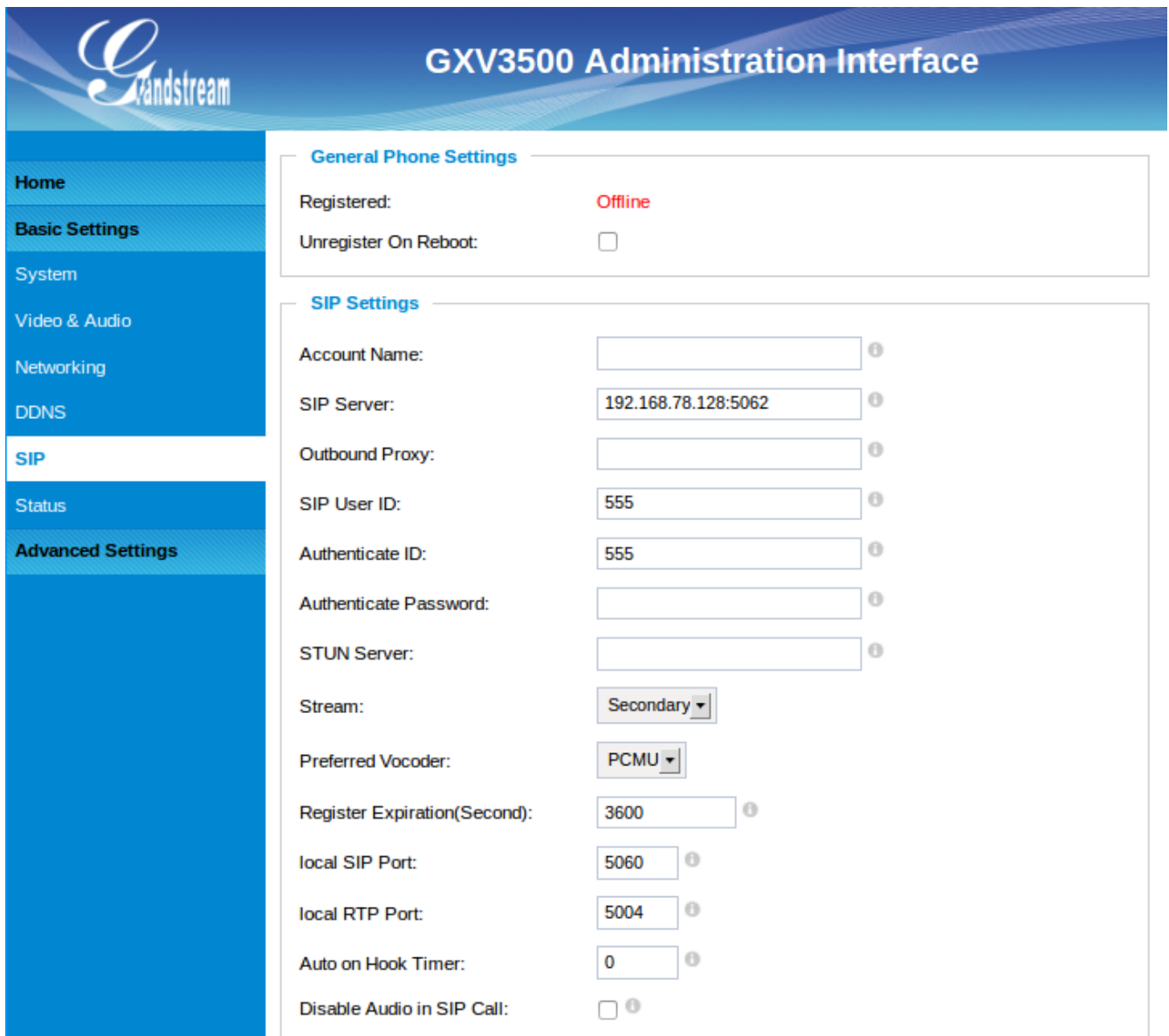
User ID	Sip Server	Sip Destination Port
<b>Unconditional Call Forward to VOIP:</b> <input type="text" value="555"/>	<input type="text" value="@ 192.168.78.126"/>	<input type="text" value="5060"/>

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## Configurações do GXV-3500:

### Aba SIP

1. Campo **“SIP Server”**: IP do HT-503 **com a porta 5062**
2. Campo **“Preferred Codec”**: PCMU



The screenshot displays the 'GXV3500 Administration Interface' with a sidebar on the left containing navigation options: Home, Basic Settings, System, Video & Audio, Networking, DDNS, SIP (highlighted), Status, and Advanced Settings. The main content area is divided into two sections: 'General Phone Settings' and 'SIP Settings'. In the 'General Phone Settings' section, 'Registered:' is shown as 'Offline' and 'Unregister On Reboot:' has an unchecked checkbox. The 'SIP Settings' section contains the following fields: 'Account Name:' (empty), 'SIP Server:' (192.168.78.128:5062), 'Outbound Proxy:' (empty), 'SIP User ID:' (555), 'Authenticate ID:' (555), 'Authenticate Password:' (empty), 'STUN Server:' (empty), 'Stream:' (Secondary), 'Preferred Vocoder:' (PCMU), 'Register Expiration(Second):' (3600), 'local SIP Port:' (5060), 'local RTP Port:' (5004), 'Auto on Hook Timer:' (0), and 'Disable Audio in SIP Call:' (unchecked).