# HT503 Hop-on/Hop-off Scenarios

## VoIP-to-PSTN Calls

This function is applicable on the FXO port that functions as a bridge between VoIP and PSTN. The user can remotely use the PSTN line to initiate a call.

### To place a VoIP-to-PSTN call:

1. Dial the FXO SIP account phone number to establish the VoIP session. The caller will hear the ring back tone once. Then the caller hears either a special continuous tone or a dial tone. The special continuous tone is played if the pin code is configured, otherwise, user will hear a dial tone
2. Enter PIN number (previously configured on the configuration page). If PIN is valid, the caller will hear a dial tone and be connected to the PSTN line.  Caller can dial number and place a local call.
3. If PIN is invalid, user is prompted to enter pin code again with a continuous tone. After 3 attempts, the HT503 will disconnect. If no PIN is entered, the HT503 will time out after 10 seconds.
4. If PIN is not configured, the system will allow all calls.  Place a call after hearing dial tone.

**Note**:

- Password protection VoIP-to-PSTN calls is optional.  A PIN consists of up to 8 numeric digits and is configured through BASIC SETTINGS of the web configuration page.  By default, there is no password protection (i.e. there is no authentication required for callers on the use of PSTN line through HT503).
- If using a PIN, the VoIP device that calls into the HT503 FXO account must be configured RFC2833 or SIP Info for DTMF digit transmission.
- During any stage of DTMF digits input, a 4 seconds timeout is applied to serve as an end of PIN or destination number input. Users may also use the “#” key to indicate the end of an input.
- On the web configuration page, if the “*Forward to PSTN*” is configured, the second stage dialing is eliminated (i.e. after dialing into the FXO SIP account number, the PSTN number will be called automatically).

## PSTN-to-VoIP Calls

This function is applicable on the FXO port that functions as a bridge between VoIP and PSTN. The user can make VoIP calls remotely by dialing into FXO line port on HT503.

### To place a PSTN-to-VoIP call:

1. Place a call from any phone (mobile or wireline) to your phone number (number is configured on the FXO port).  Your analog phone will ring 4 times by default.  You can change this setting on the configuration page.
2. After 4 rings, the caller hears either a special continuous tone or a dial tone. The continuous tone indicates a PIN is necessary; a dial tone indicates you can place a call.
3. Enter PIN number (previously configured on the configuration page). If PIN is valid, the caller will hear a dial tone and be bridged to VoIP (a SIP account must be configured on the FXO port). The caller can dial a **VoIP number** (can it be any phone number?) followed by # (or wait for 4 seconds).
4. If PIN is invalid, user is prompted to enter PIN again with a continuous tone. After 3 attempts, the HT503 will disconnect. If no PIN is entered, the HT503 will time out after 10 seconds.
5. If PIN is not configured, the system will allow all calls.  Place a call after hearing dial tone.

**Note:**

- Password protection VoIP-to-PSTN calls is optional. A PIN f consists of up to 8 numeric digits and is configured through BASIC SETTINGS of the web configuration page. By default, there is no password protection (i.e. there is no authentication required for VoIP SIP account on FXO port).
- If using a PIN, the VoIP device that calls into the HT503 FXO account must be configured RFC2833 or SIP Info for DTMF digit transmission.
- During any stage of DTMF digits input, a 4 seconds timeout is applied to serve as an end of PIN or destination number input. Users may also use the "#" key to indicate the end of an input.
- On the web configuration page, if the "*Forward to PSTN*" is configured, the second stage dialing is eliminated (i.e. after bridging to VoIP, the configured VoIP number will be called automatically).

## Route Calls to PSTN

This function is applicable on the FXO port that can access the PSTN network. By default, HT503 is in VoIP mode when off-hook. This call feature is especially useful for emergency calls or to place local telephone calls.

To use this feature, specify a prefix or a telephone number in the "*Route call to PSTN*" on BASIC SETTINGS web configuration page. This will route specific calls over the FXO PSTN line port (local PSTN network). If the dialed digits match one of the specified prefixes, outbound calls will be initiated from the PSTN line (i.e. if "Route call to PSTN" is configured to be 626, all outgoing calls start with 626 will be initiated from PSTN line).