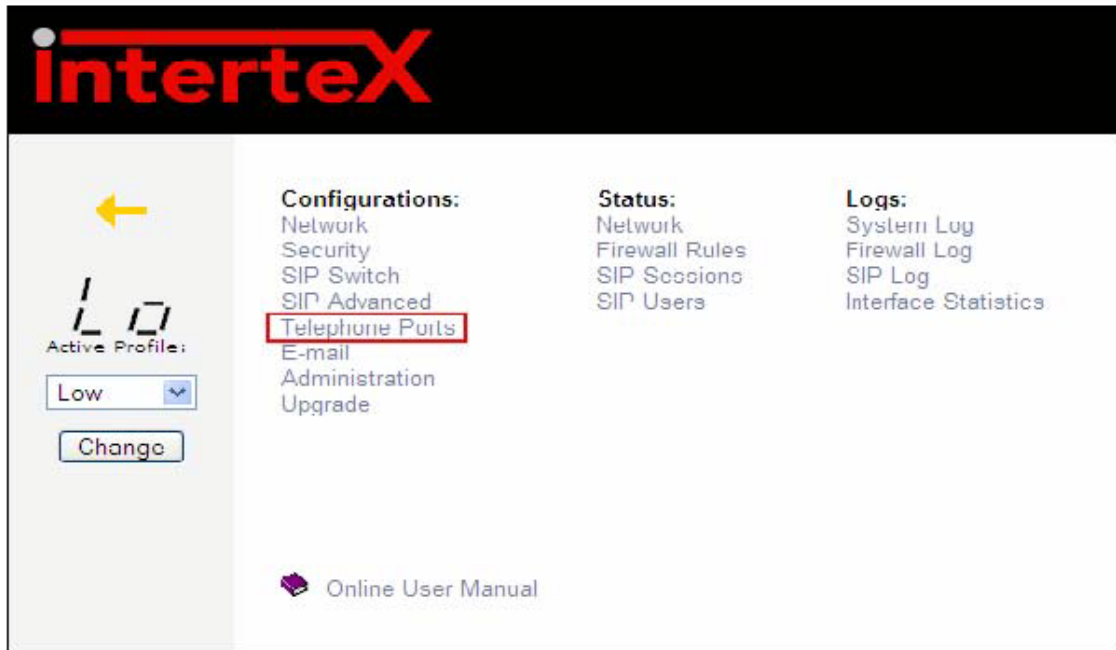


Configuring Intertex SurfinBird IX67 FW AIR GW2 for PSTN Failover

1. Configure the Router based on the Worldsmart-Intertex configuration guide.
2. Connect Telephone Line to the Router's LINE Port (FXO).
3. Connect a Telephone to the Router's PHONE Port 1 (FXS). The router has 2 such ports; you can connect 2 phones to the router, if required. This document describes phone configuration only on the first FXS port.
4. Access the Router's web configuration page, and login.
5. Click on the *Telephone Ports* Link. This will take you to the Telephone Ports configuration page.



6. Go to the Phone Port 1 (FXS 1) Section. Configure the Phone Port. Enter a username. The picture below shows *pstn-line* as the username. The phone connected to this port will be registered with the router itself (unless you specify a full SIP address).

Phone Port 1 (FXS 1) SIP Registration ?

SIP address: (e.g. phone@smartcompany.com) Display name: (Optional)

Authentication: User ID: Password:

Call Control Phone 1 ?

Enter number or URI:

Multiple calls:

Configuration:

- Call Waiting (#430 to enable, #431 to disable)
- Do not disturb (#730 to enable, #731 to disable)
- Call Anonymously (#630 to enable, #631 to disable, #67 to enable for next call)
- Telephony Keys Style:

7. Go to the Telephone Line Port (FXO) Section. Enter a domain name. Default is *localgw*. The example below shows *myoffice* as the domain name.
8. Add a dialing prefix for the outgoing calls. (** is the default).
9. If you want to forward all calls coming through the telephone line to the phone connected to the PHONE port1, add the username *pstn-line* in the field provided as shown below. You can also specify the number of ring signals for this. The router also provides you an option to forward these calls to a SIP phone that is registered with a SIP provider. See the below image.

Telephone Line Port (FXO) ?

Domain name: (Dial using <telephone number>@<domain name>) Add a domain name

Telephone number: Display name: (Optional)

Authentication: User ID: Password: (Optional)

Outgoing calls

Allowed to call out: Inside users: Outside users:

Await dial tone after initially dialed: (Enter 0 or 9 if behind PBX)

Prefix for dialing through this port: (If "***" then dial using **<telephone number>) Set Dial Prefix
(Will be overridden if Dial Plan in SIP Switch is used)

Incoming calls

Forward incoming calls to: during ring signals Incoming calls through the phone line

Collect: extension digits and forward to SIP server @ (Built-in server if empty)

Try forwarding to collected extension for: seconds

Forward to: after timeout or failure above Forward to another SIP phone



How it works

Outbound Calls:

The above setup implements Intertex-PSTN integration. SIP phones that are behind this router, registered with Worldsmart server will function normally. Calls made from these phones will hit the Worldsmart server and will be routed to the desired destination.

However whenever a number is dialed with **prefix, or dialed with the domain name in the host part, these will be routed through the telephone line port.

For example:

- Dial 18008050558 – SIP call, this will hit the Worldsmart server
- Dial 18008050558@myoffice – PSTN call, call will go directly to PSTN through the telephone line port.
- Dial **18008050558 – PSTN call, call will go directly to PSTN through the telephone line port.

Inbound Calls:

Based on the configuration described in this document, whenever an inbound PSTN call reaches the router through the telephone line, it will be routed to the phone connected to the first PHONE port (pstn-line). As you can also see, there is a timeout period of 10 seconds set for the inbound calls. If the call is not answered within 10 seconds, it will be forwarded to a SIP phone – gopinath@iptel.org in this case – registered with a different SIP provider.

Notes:

Call Disconnect:

The inbound calls that are coming through the telephone line do not terminate properly, when they are answered on the phone connected to the FXS port. Disconnecting at one end does not disconnect the other. You have to explicitly disconnect such calls at both ends. I am awaiting Intertex Engineering's response about this.

Caller-ID:

The Caller-ID feature for inbound calls coming through the telephone line is OFF by default on this router. This is because; this feature is in Beta state. You need to enable it with either **V.23** or **Bell 202** options (based on what the PSTN operator is using). Note that you will have to make several inbound calls before the router identifies the Caller-ID for the first time. After that you should see consistent results. I have achieved 90% accuracy, after the router identified the Caller-ID for the first time.

To set the Caller-ID, click on the *Advanced Telephone Port* options on the Telephone Port page. Configure it as shown in the below pictures:

Advanced Settings for Telephone Ports



Phone Port 1 (FXS 1) ?

Ring signal	<input type="text" value="Normal"/>	▼
Dialling timeout	<input type="text" value="4"/>	seconds, for each digit
Maximum call duration	<input type="text" value="0"/>	seconds (0 means infinite)
Disconnect call if silent for	<input type="text" value="0"/>	seconds (0 means infinite)
Disconnect call if no RTP received for	<input type="text" value="3600"/>	seconds (0 means infinite)
Voice codecs used	<input type="text" value="Prioritize quality, accept high bitrate: G.711, G.729"/> ▼	
	<input checked="" type="checkbox"/>	Adapt to remote side codec prioritization
	<input type="checkbox"/>	Select a single codec for incoming calls
Desired voice packet time	<input type="text" value="Codec default"/> ▼	
DTMF transmission	<input type="text" value="Encoded as events (out-of-band)"/> ▼	
Volume: Speaker	<input type="text" value="0"/>	(-20...+20 dB)
Volume: Microphone	<input type="text" value="0"/>	(-20...+20 dB)
Silence suppression	<input type="text" value="Off"/> ▼	
Echo cancellation	<input type="text" value="Non-linear with comfort noise generation"/> ▼	
Caller ID mode	<input type="text" value="Bell 202"/> ▼	Caller ID
Caller ID number presentation	<input type="text" value="Profile Default"/> ▼	
Caller ID name presentation	<input type="text" value="Profile Default"/> ▼	

Telephone Line Port (FXO)



Codec sharing with FXS	<input type="text" value="FXO-FXS loop preferred, allow usage of any FXS codec"/>
Extension dialling timeout	<input type="text" value="4"/> seconds, for each digit
Maximum call duration	<input type="text" value="3600"/> seconds (0 means infinite)
Disconnect call if silent for	<input type="text" value="30"/> seconds (0 means infinite)
Disconnect call if no RTP received for	<input type="text" value="300"/> seconds (0 means infinite)
Voice codecs used	<input type="text" value="Prioritize quality, accept high bitrate: G.711, G.729"/>
	<input checked="" type="checkbox"/> Adapt to remote side codec prioritization
	<input type="checkbox"/> Select a single codec for incoming calls
Desired voice packet time	<input type="text" value="Codec default"/>
DTMF transmission	<input type="text" value="Encoded as events (out-of-band)"/>
Volume: Line in	<input type="text" value="0"/> (-20...+20 dB)
Volume: Line out	<input type="text" value="0"/> (-20...+20 dB)
Silence suppression	<input type="text" value="Codec standard"/>
Echo cancellation	<input type="text" value="Non-linear with comfort noise generation"/>
Receive Caller ID	<input type="text" value="Bell 202"/> Caller ID