

Connecting Sipura ATAs to a legacy PBX System.

The FXO port (line port) of a Sipura SPA-3000 can be connected up to an analogue extension port of a PBX system in just the same way it is connected to an analogue phone line. This means that you can use Sipura adaptors as a long-line extension to a PBX as rather than an analogue telephone line as they are more commonly used with.

The SPA-3000 FXO port requires an RJ-11 plug wired to the US-specification for analogue telephone wiring with the centre two wires (red and green) as the ring/tip. See the diagram below:

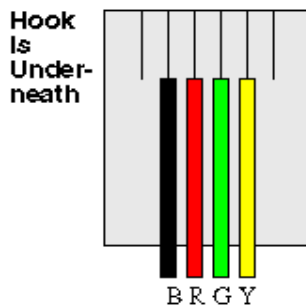


Diagram: US-Spec RJ-11 Plug Wiring

Most PBXs in the UK will use either RJ-11 wiring as above or BT-style wiring with plugs as the diagram below, these use pins 5 & 2 for ring/tip:



Diagram: BT-style Telephone Plug

In this case ProVu can supply RJ-11 to BT adaptors. You should use the black RJ-11 cable supplied with the SPA-3000 along with the adaptor supplied by ProVu. Please note, not all adaptors are the same so we recommend you buy from us to make sure you get the right adaptor.

In some cases a PBX system can use a different type of cabling, sometimes RJ-45 (like computer network cabling) or some other proprietary system. In these cases you will need to contact your telephone engineer or PBX supplier to find out what the wiring is like & how to make an adaptor to suit. This is something only your PBX manufacturer or telephone engineer will know how to do. Also note that the Sipura adaptors can only accept an analogue connection, some PBXs have digital connections & these will not work with the Sipura adaptor, your PBX must have an analogue extension port.

One of the biggest stumbling blocks in getting this to work is the behaviour of the extension port on the PBX. Unfortunately these often don't behave like a normal BT line. Quite often they will use non-BT-standard voltages to signify on hook, ringing and off hook. Another possibility is non-BT-standard call progress tones, most of which won't actually matter except the disconnect tone which can cause calls to continue even when the phone is placed back on hook. The SPA-3000 can be configured to work with just about any PBX system but it will require some knowledge of how your particular PBX behaves. This may come from a manual supplied with



the PBX or from the engineer who originally installed it.

Since we don't deal with analogue PBX systems, ProVu cannot assist with wiring configurations, determining line voltages or progress tones. Other than what has been said in this document.

In addition to the SPA-3000 connected to the PBX system you will also need a further adaptor or a Sipura IP phone (e.g. The SPA-841) at the remote end (e.g. home worker's house).

How To Setup the Extension

There are two main ways of setting up the extension, the first uses only IP addresses to communicate between the two units. You will need to have each adaptor on a static IP address with no NATs in-between, ideally a VPN would be setup between the two units. How to setup a VPN is beyond the scope of this document but there is lots of information on the Internet. If you don't have a VPN & have the Sipuras on private IP addresses then the best option is to subscribe with a VoIP service provider, there are plenty of these who offer very cheap or even free accounts for VoIP communication only which is all you need.

IP Addressing

SPA3000 Setup:

Login to the web interface by typing it's IP address into a PC connected to the same subnet & click "admin login" & "advanced".

Go into the tab for the "PSTN Line" & alter the following settings:

- "Make Call Without Reg" to "yes".
- "Ans Call Without Reg" to "yes".
- Change dialplan 2 to read "(S0<:81.187.74.25>)". This sets up a hotline which calls the remote SPA using it's IP address at the standard SIP port of 5060. It's wise to leave use dialplan 2 (or any number after that) & not dialplan 1 since most of the default settings on this page are set to use dialplan 1.
- "PSTN Ring Thru Line 1" if you have a phone connected to the 3000 & want it to ring when a call is received via the PBX, leave this as "yes". If you only want the phone on connected to the remote SPA to ring, set this to "no".
- "PSTN Caller Default DP" to "2". This should match whichever dialplan you setup two steps ago. Again, try to avoid changing dialplan 1 in this case.
- "PSTN Answer Delay" change this to zero. Otherwise any incoming calls will not be forwarded to the remote SPA for 16 seconds (as the default value here is 16).
- "Line 1 VoIP Caller DP" to "none".
- "VoIP Caller Default DP" to "none".

In the "Line 1" tab, change the following settings:

- "Make Call Without Reg" to "yes".
- "Ans Call Without Reg" to "yes".
- "Enable IP Dialling" to "yes".

Remote SPA Setup:

In the web interface go to the tab for "line 1/Ext 1" and alter the following settings:



- “Make Call Without Reg” to “yes”. This allows calls to be made without being registered to a SIP registrar.
- “Ans Call Without Reg” to “yes”. This allows calls to be received without being registered.
- “Enable IP Dialling” to “yes”. This enables the SPA to dial using IP addresses rather than SIP URIs which is ideally what should be used.
- Change the dial plan to read “(S0<:81.187.74.26:5061>)”. This sets up what is called a “hotline”. So when the phone is picked up, it automatically dials the number (or IP address in this case) without the user doing anything. Here it connects to the first SPA3000 using port 5061 which is where the FXO port is listening for incoming requests.

Security:

What has been done so far will work but so far anyone could type the IP address of the SPA3000 into an IP phone & make calls through your PSTN or PBX!

Alter these settings in the “PSTN Line” tab of the SPA3000:

- “VoIP Caller Auth Method” change to “PIN”. This means that anyone accessing your PSTN line will be confronted by a set of beeps indicating that they should enter a PIN number. Failure to do so means they cannot access your PBX.
- “VoIP Access List” add the IP address of the remote SPA into here: 81.187.74.25 in this case. This allows your remote SPA to bypass the PIN number authorisation.
- “VoIP Caller 1 PIN” change this to any PIN number you want. This is what people will have to enter in to access the PBX.

Whilst stopping unauthorised users accessing your PSTN line this also means you can call your SPA3000's IP address from any IP phone & make calls using your own PBX once you've entered your PIN number in!

Using SIP Accounts.

IP addressing between the Sipura units will not work well (if at all) when there are NATs in between & the Sipuras are on private IP addresses (addresses beginning with “192.168.”, “10.0.” or “172.16.”). In this case the easiest solution is to open an account with a VoIP service provider who can give you:

- Two SIP addresses with passwords.
- Proxy server name.
- Outbound Proxy server name.

The Outbound proxy server is where the NAT traversal is achieved. Some service providers use STUN but we find Outbound proxies (sometimes called Session Border Controllers) work more reliably.

In this example the address “sip1@proxy.com” will be for the SPA-3000, “sip2@proxy.com” will be for the remote SPA. The Proxy server will be “proxy.com” & the Outbound proxy will be “nat.proxy.com”. Replace these values with your own service provider's details.

SPA3000 Setup:

Login to the web interface by typing its IP address into a PC connected to the same subnet & click “admin login” & “advanced”.



Go into the tab for the "PSTN Line" & alter the following settings:

- Change dialplan 2 to read "(S0<:sip2@proxy.com>)".
- "PSTN Ring Thru Line 1" if you have a phone connected to the 3000 & want it to ring when a call is received via the PBX, leave this as "yes". If you only want the phone on connected to the remote SPA to ring, set this to "no".
- "PSTN Caller Default DP" to "2". This should match whichever dialplan you setup two steps ago.
- "PSTN Answer Delay" change this to zero. Otherwise any incoming calls will not be forwarded to the remote SPA for 16 seconds (as the default value here is 16).
- "Line 1 VoIP Caller DP" to "none".
- "VoIP Caller Default DP" to "none".
- "Proxy" to "proxy.com".
- "Outbound Proxy" to "nat.proxy.com".
- "Use Outbound Proxy" to "yes".
- "User ID" to "sip1".
- "Password" to the password provided by your service provider.
- "Auth ID" to "sip1".
- "Use Auth ID" to "yes".

Remote SPA Setup:

In the web interface go to the tab for "line 1/Ext 1" and alter the following settings:

- Change the dial plan to read "(S0<:sip1@proxy.com>)".
- "Proxy" to "proxy.com".
- "Outbound Proxy" to "nat.proxy.com".
- "Use Outbound Proxy" to "yes".
- "User ID" to "sip2".
- "Password" to the password provided by your service provider.
- "Auth ID" to "sip2".
- "Use Auth ID" to "yes".

Note: the "Auth ID" setting isn't used by all service providers but is sometimes required. If you are having trouble getting the Sipuras to register, try removing that setting & change "Use Auth ID" back to "no".

Security:

What has been done so far will work but so far anyone could type the IP address of the SPA3000 into an IP phone & make calls through your PSTN or PBX!

Alter these settings in the "PSTN Line" tab of the SPA3000:

- "VoIP Caller Auth Method" change to "PIN". This means that anyone accessing your PSTN line will be confronted by a set of beeps indicating that they should enter a PIN number. Failure to do so means they cannot access your PBX.



- “VoIP Access List” add the SIP address of the remote SPA into here: sip2@proxy.com in this case. This allows your remote SPA to bypass the PIN number authorisation.
- “VoIP Caller 1 PIN” change this to any PIN number you want. This is what people will have to enter in to access the PBX.

Troubleshooting

At this point you should be able to pickup the handset of the remote Sipura & hear the dialtone from the PBX system. You should also be able to make & receive calls from it. However, there may be a few “bugs” that need ironing out & these are specific to the PBX you are using so it's usually necessary to contact your telephone administrator or PBX supplier for more information.

Problem: When I hang up the telephone on the far end of the extension, the PBX doesn't clear down the call (doesn't hang up).

Solution: The Sipura 3000 is capable of generating disconnect signals in various ways. The most common is to generate a disconnect tone to be recognised by the PBX. The Sipura comes with the American disconnect tone which may or may not be suitable. You will have to find out how your PBX deals with disconnect detection, if it is a tone you need to know the frequencies involved & the cadence. When you have these values, consult the Sipura SPA Admin Guide which can be found on the Support section of <http://www.provu.com>. The disconnect tone is entered in on the PSTN Line page of the SPA-3000 web interface.

Problem: Caller ID doesn't work.

Solution: You will need to contact your telephone administrator or PBX supplier to find out what standard or caller ID your PBX generates. The American standard is Bellcore & the UK standard is ETSI FSK with PR but your PBX could use any type. The setting you need to change is on the “Regional” tab of the SPA-3000 web interface, called “Caller ID Method”.