

Set up call routing, forwarding and routing features

Accessing the routing wizard

All call forwarding and routing features are accessible via the routing wizard. Each number has its own routing wizard. To access the routing wizard for any of your numbers

- Log in at <http://voip.sapien.it>
- Click on My Services >> My Numbers
- Click on the number that you're interested in
- About half way down the page, you'll see a button labelled routing wizard. The list in the box under this button will tell you how your number is currently routed
- Click on the routing wizard button to edit the routing for your number

How the routing wizard works

The routing wizard works in steps. It will start by asking you what you want to do with your call in the first step; for example, you might want to play a welcome message to your caller.

Each time you define a new step and save it, the routing wizard will ask you what you want to do in the next step. It will keep asking you for steps until you define a step that will finish the routing. The only two steps that will finish routing are hang up or send to voicemail.

This can be a little confusing at first, but if you remember that it will only do the final step if the call has not been answered in any of the previous steps then it should make sense.

Routing steps

You can define as many steps as you want, in each step you can:

- Forward the call to a single external number, like a mobile or landline
- Forward the call to a SIP url (this is another VoIP address)
- Send the call to your IP phone or a group of IP phones
- Send the call to a group made up of any combination of VoIP phones, mobiles, landlines or SIP addresses
- Send the call to voicemail / voicemail to email
- Play a greeting to your caller
- Play music whilst ringing the call
- Hangup the call

At each step, you define how long you want that step to ring for. For example, you might want to send the call to your home landline for 30 seconds and then go to voicemail if you don't answer. In this case you would define:

- Step 1: Ring my landline [number] for 30 seconds
- Step 2: Go to voicemail

Note that the routing wizard will skip a step if all the devices it tries to ring are busy in any step. So in the above example, if your home phone is busy, then it would skip step 1 and go straight to voicemail.

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Using Voicemail

Accessing your voicemail

You can access your voicemail from an IP phone connected to Sapien IT over VoIP by dialling 8000. This will take you to the main voicemail menu. No PIN is required in this case, as the device is already authenticated

To access your voicemail externally, you need to dial the external telephone number of your business extension. Let it ring to voicemail and then press the star(*) key. You will be prompted for your PIN number. The default for your PIN number is the first four digits of the VoIP username, seen on the My numbers page on the VoIP control panel. So for example if your VoIP username is 100000000 then your default Voicemail PIN will be 1000

You can change your PIN number in the Mailbox options menu (item 0) of your voicemail menu. See below.

Changing your voicemail greeting

Access you voicemail as described above and then choose option 0 followed by option 4 (record your unavailable message). You should then be prompted to re-record your greeting.

Sending calls to voicemail

In order to send a call to voicemail, no special configuration or settings are needed on your phone. Simply choose the 'send call to voicemail' step in the routing wizard for your number.

For example, you might set your routing up like this: 'Send the call to my ip phone for 30 seconds and then if busy or no answer go to voicemail'. So if your phone is busy, or you don't answer, or you click reject on your phone, the call will go to voicemail.

Voicemail to email

If you've set the email account on your number for voicemail then we'll also deliver your voicemail by email too as a .wav (wave file) attachment.

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The voicemail menu

The voicemail menu allows you to perform a number of tasks in addition to listening to new messages. This menu is accessible as described above. The menu structure is defined below.

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+-1 Read voicemail messages
| +-3 Advanced options
| | +-1 Reply
| | +-2 Call back(1)
| | +-3 Envelope
| | +-4 Outgoing call(1)
| | +-5 Send Message
| +-4 Play previous message
| +-5 Repeat current message
| +-6 Play next message
| +-7 Delete current message
| +-8 Forward message to another mailbox
| | +-1 Use Voicemailnumber
| +-9 Save message in a folder
| | +-0 Save in new Messages
| | +-1 Save in old Messages
| | +-2 Save in Work Messages
| | +-3 Save in Family Messages
| | +-4 Save in Friends Messages
| | +-* Help; during msg playback: Rewind
| | +-# Exit; during msg playback: Skip forward
+-2 Change folders
| +-0 Switch to new Messages
| +-1 Switch to old Messages
| +-2 Switch to Work Messages
| +-3 Switch to Family Messages
| +-4 Switch to Friends Messages
+-3 Advanced Options
| +-5 Send Message
| | +-1 Use Voicemailnumber
+-0 Mailbox options
| +-3 Record your name
| +-4 Record your unavailable message
| | +-1 Record your unavailable message
| | +-2 Erase your unavailable message (going back to the standard message)
| +-5 Change your password
| +-* Return to the main menu
+-* Help
+-# Exit

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VoIP User Guide

The audio quality of my calls is poor or my calls keep dropping

Common causes of audio quality issues

When supporting customers, we find that the most common cause of call audio quality issues is the quality and capacity of the customers' Internet connection.

This is especially true if you find that you can hear your callers fine, but they sometimes struggle to hear you.

Symptoms of broadband/ Internet connection issues are:

- Caller complaining of audio breaking up or jittery
- Calls dropping at random points during the call
- Long periods of no audio or audio being lost completely
- Lots of delay on the call - in either direction

Other causes of dropped calls

Dropped calls in particular can be caused by issues other than broadband connection quality. On occasion, routers can block some of our traffic and we've included an article below on how to change your port numbers to avoid this happening.

How to check your connection

You can check the quality and capacity of your connection pretty easily using two external resources:

- speedtest.net can help you to perform a speed test on your line to inform you of what sort of connection speed you have. You need a minimum of 150kbps (that's 0.15 Mbps) up and down for a single VoIP phone and an additional 120kbps (0.12 Mbps) for each additional phone you have. Pay particular attention to the upstream value reported
- pingtest.net tests the quality of your connection rather than just the speed. You really need a grade B or above connection for a good VoIP service

Try to run tests on both of these sites at points in the day when you are experiencing the problem.

Other tests

If your tests come back ok, or you need any other advice, then get in touch with us: we can run a 24 hours trace on your Internet connection; which will give us some more specific data on the quality of the route between your ISP and our servers.