

# Swyx PBX - Connect to jtel System

## Background

This guide describes one way of connecting the jtel ACD to the Swyx PBX system and routing calls to the jtel System.

The following principle applies to connecting the jtel System to the Swyx PBX:

- Incoming calls to the jtel System will be routed by SIP with RTP to the jtel ACD on Leg 1 by the PBX
- The jtel ACD will make outgoing calls to agents via SIP with RTP back down the same route back to the PBX on Leg 2
- For outbound, Leg 1 is to the agent first and Leg 2 to the destination.

The jtel System hence operates as a back to back user agent, the Swyx PBX is not aware that Leg 2 calls are associated with Leg 1.

## Scenario

The following configuration scenario was performed after the following steps had been performed on the Swyx System:

- Install Swyx
- Get System up and running
- Configure and test internal users
  - Check telephony between users
- Configure a trunk to the outside world
  - Check outgoing calls from users
- Configure routing to internal users from outside
  - Check incoming calls via the trunk to internal users

After these steps, a basic PBX is configured with some users, internal calls working, and functionality to call inbound or outbound to / from the outside world.

### Caution



This is not intended to be an expert guide on configuring the Swyx PBX system or a recommendation that "this is the one and only way" of doing things.

You should know your Swyx PBX well, and any configuration therein which is relevant.

You should also be familiar with the procedures involved here, particularly in the Swyx PBX.

You should also be capable of identifying steps which may be critical to your PBX installation, it's configuration or any routing involved, before you proceed.

The scenario below may need modifying to suit your needs!

## First step: Add Trunk Group for jtel ACD

### Add new trunk group

- Right mouse click

The screenshot displays the SwyxWare Administration console interface. On the left, a tree view shows the hierarchy: Console Root > SwyxWare Administration > SwyxServer TEST-SWYX > Trunk Groups > Trunks. A context menu is open over the 'Trunks' node, listing options: 'Add Trunk Group...' (highlighted in blue), 'Neues Fenster hier öffnen', 'Aktualisieren', and 'Hilfe'. On the right, a details pane titled 'Name' shows the value 'SwyxServer TEST-SWYX'.

**Go to next step**



**Specify a trunk group name and a description**

Add new Trunk Group ✕

**Trunk Group Name and Description**  
Specify Trunk Group name and description. 

Enter a unique Trunk Group name, i.e. not used otherwise as Trunk name, User name, Group name or Phonebook entry.

Enter the optional description that will later on help you identifying this Trunk Group.

Trunk Group Name:

Description:

## Trunk Group Type

- It is important to specify the trunk group type:
  - trunk Group type : **SIP**
  - profile: **<Customized SIP>**

Add new Trunk Group ✕

**Trunk Group Type**  
Specify the type of the Trunk Group and select the appropriate profile. 

Select the Type of Trunk Group to be added from the first list and choose the applicable profile from the second list. If you are uncertain, which profile is applicable for your installation, consult the SwyxWare Administration documentation.

If you want to add a Trunk Group for a non-listed SIP service provider, select the Profile "Custom". This will allow entering all required parameters in subsequent steps.

Trunk Group Type:

Profile:

## SIP Settings

- Disable sip registration and go to next step

Add new Trunk Group ×

**SIP settings** 

Please specify whether SIP registration is enabled for this Trunk Group.

The subsequently prompted information must have been supplied by your SIP service provider.

If your service provider requires a SIP registration (usual case), enable the checkmark and enter the registrar's name or IP address.

The SIP account specific information must be entered when you add a Trunk to the Trunk Group you are currently creating.

Enable SIP registration

Registrar:  :  

Re-registration Interval:    seconds

## SIP Settings

- The proxy address is the IP Address of the jtel System telephony server
- Specify port 5060 as standard (unless this has been changed deliberately on the jtel System)
- For the realm specify the ip address
- DTMF Mode = RFC 2833 Event

Add new Trunk Group ✕

**SIP Settings**  
Specify SIP settings for this Trunk Group. 

The SIP Proxy is the service provider's interface for call control. Therefore its name or IP address must have been provided.

The SIP realm is part of the SIP addressing mechanism, i.e. it is used for SIP URI composition. The parameter "DTMF Mode" determines how a user's keypad input is passed to the provider.

Proxy:  :  

Realm:

DTMF Mode:

## Stun Server Settings

- Disable STUN

Add new Trunk Group ×

**STUN Server Settings**  
Specify STUN Server Settings. 

A STUN server can be used to traverse non-symmetric NAT firewalls, in order to access another SIP proxy. The STUN server must be located in the public Internet.

Please enter the name or IP address of the STUN server and the STUN service port (usually 3478). A publicly available STUN server is e.g. "stunserver.org".

Enable STUN support

STUN Server:  Port:

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## Encryption Settings

- choose udp as a transport protocol

Add new Trunk Group ×

**Encryption Settings** 

Please specify the SIP Transport Protocol and the Voice Encryption Mode for this Trunk Group.

Some SIP providers require a specific transport protocol. If you choose "Automatic", the transport protocol will be determined via DNS resolution.

Voice Encryption can only be configured, if "TLS" is selected as transport protocol.

Transport Protocol:

Encryption Mode:

## Definition of Routing

- Choose "Do not create initial Routing Records"

Add new Trunk Group ×

**Definition of Routing** 

Specify for what calls this Trunk Group is supposed to be used.

Depending on your choice, initial Routing Records will be created.  
Public Numbers should be added in canonical format (e.g. "+4930123456"), "\*" can be used as a wildcard.

Use Trunks of this Trunk Group...

for all external calls

for all external calls to the following Called Party Number or SIP URI only:

for all external calls and all unassigned Internal Numbers

for Internal Numbers:

Do not create initial Routing Records

## Location Profile

- Application location profile : keep it to Default Location

Add new Trunk Group ×

**Location Profile**   
Select the applicable Location Profile for this Trunk Group.

A Location within SwyxWare defines all location specific settings like the time zone, the required public access code, the country and area codes.

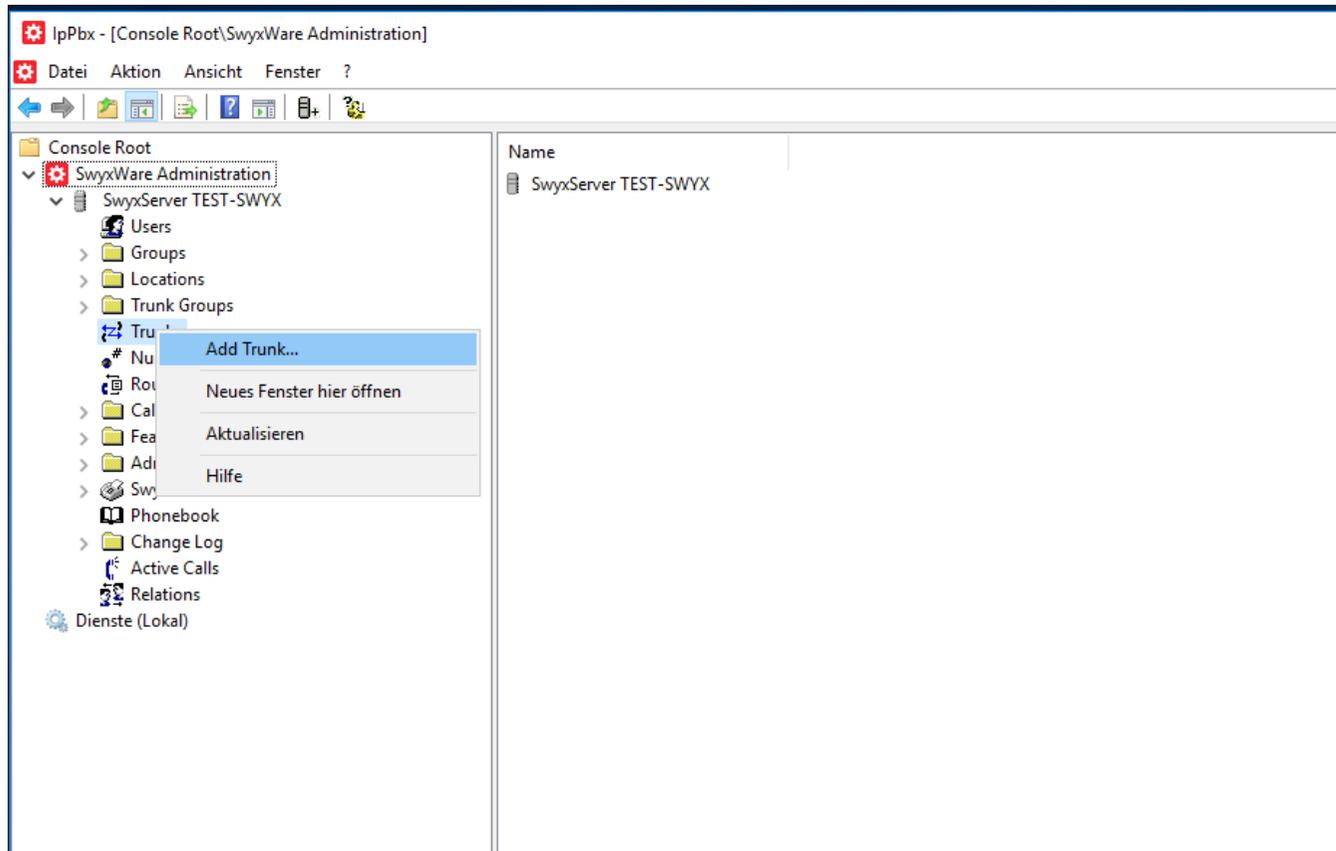
Please select one of the listed Locations which will be assigned to this Trunk Group.

Location:

Description

Second step: Add Trunk for jtel ACD

Add new trunk



**Go to next step**



### Trunk Name

- Choose a trunk name and a description

Add new Trunk ×

**Trunk Name**  
Choose an unique name for the new Trunk. 

Enter a unique Trunk name, i.e. not used otherwise as Trunk Group name, User name, Group name or Phonebook entry.

Enter the optional description that will later on help you identifying this Trunk.

Trunk Name:

Description:

### Trunk Group Selection

- choose the the appropriate trunk group created above

Add new Trunk ×

**Trunk Group Selection** 

Assign the new Trunk to an existing Trunk Group or create a new Trunk Group to which the new Trunk will be assigned.

The chosen Trunk Group determines the type of Trunk (ISDN/Analogue/SIP Gateway, SIP or ENUM Trunk, SwyxLink) and defines several common properties.

Furthermore Trunk Group settings specify if a Trunk is considered for routing outbound calls.

Trunk Group:  ▼

### SIP Trunk Provider / User Data

- No special settings required - go to the next step

Add new Trunk ×

**SIP Trunk Provider / User Data**   
Specify your account data.

Enter the user identification data as provided by your SIP service provider. The user ID will be used to compose your SIP address while user name and password will be used for authentication.

SIP Provider:

User ID:

User Name:

Password:

Repeat Password:

## Subscriber Numbers

- Set all fields empty

Add new Trunk ×

**Subscriber Numbers**  
Specify Subscriber Numbers. 

Enter the subscriber number part of the Public Numbers that are terminated by this Trunk.  
If your set of subscriber numbers is incoherent enter only the first subscriber number and add the other subscriber numbers later via the Trunk's properties.  
If this Trunk does not add any Public Numbers to the system, leave all fields empty and click 'Next'.  
Note: Country Code and Area Code have been pre-determined by the Trunk Group's location.

Country Code	Area Code	First Subscriber Number	Last Subscriber Number
<input type="text"/>	<input type="text"/>	<input type="text"/>	<input type="text"/>

## SIP URI

- Set all fields empty

Add new Trunk ✕

**SIP URI**  
Specify SIP URI. 

If this Trunk is supposed to handle non-numeric SIP URIs (e.g. assigned by your SIP service provider) you can enter one of these bellow and add other URIs later via the Trunk's properties.

SIP URIs have the following format:

sip:<name1> @ <name2>

with <name1> reflecting the user's name and <name2> the realm.

For convenient input "\*" can be used as wildcard so that \*@company.com would address all users in the realm "company.com". The realm field shown below is pre-filled with the configured realm in the SIP properties but may be overwritten case by case.

URI:            sip:  @

## Codecs

- Leave as default

Add new Trunk X

**Codecs** 

Select the codecs to be used for data transmission.

The selected codec preference and filter defines the type of compression for calls using this Trunk. Therefore the selected codec has an impact on the used bandwidth and the quality of the call.

Codecs Preference and Filter

Prefer Quality ▼

- G.722 (approx. 84 kBit/s per call)
- G.711a (approx. 84 kBit/s per call)
- G.711μ (approx. 84 kBit/s per call)
- G.729 (approx. 24 kBit/s per call)
- Fax over IP (T.38, approx. 20 kBit/s per call)

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## Number of Channels

- Set number of simultaneous call to 4 (just for testing), or to the number of licensed channels (see your jtel ACD configuration - number of licensed SIP / RTP channels).

Add new Trunk ✕

**Number of Channels**  
Select number of Channels to be used by this Trunk. 

The number of concurrent calls via a specific Trunk is usually limited by the Trunk's physics, the available bandwidth or by a provider limitation.

Furthermore the number of simultaneous calls can artificially be limited to reserve (e.g. ISDN) channels or bandwidth for other applications.

Usually ISDN BRI interfaces would allow to make up to 2 simultaneous calls, while ISDN PRI interfaces allow up to 30 calls.

Number of simultaneous calls on this Trunk:

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## Computer Name

- set the name of the computer where you run swyx

Add new Trunk ×

**Computer Name**  
Define the computer name where the Trunk is hosted. 

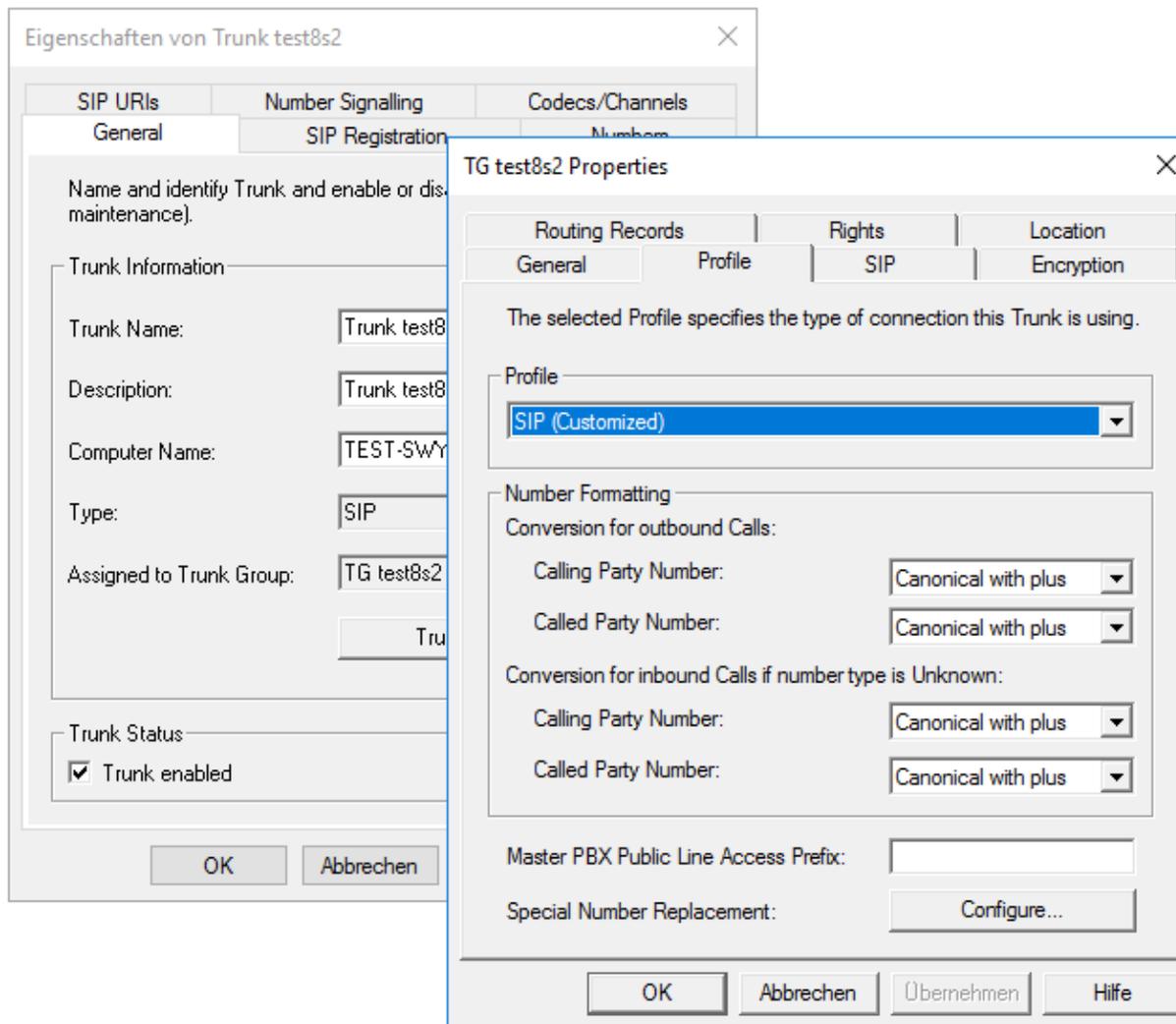
The Trunk may be hosted on another computer than the SwyxServer. In this case, the computer name must be provided here, otherwise keep the proposed default.

Please enter the computer name as it is given in the Windows Server's system properties.

Computer Name:

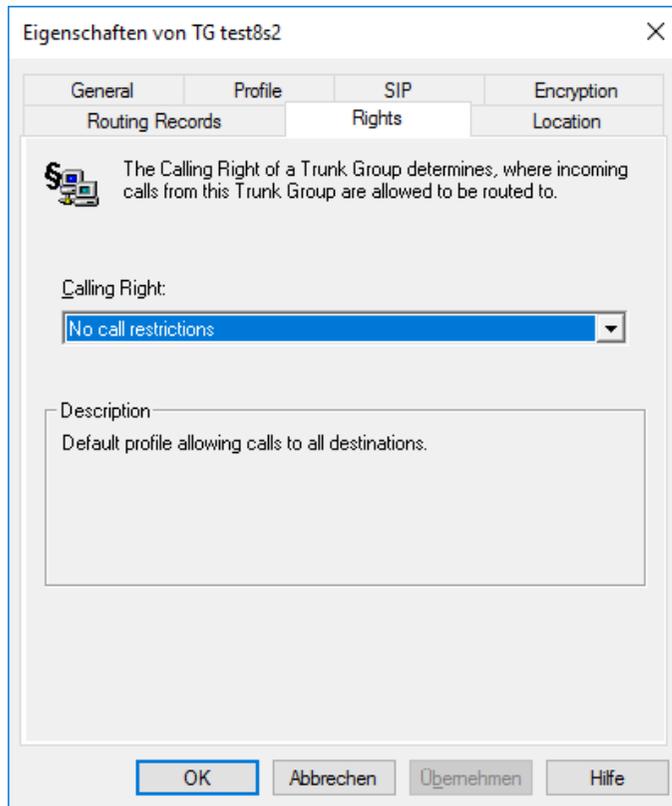
### Trunk - Trunk Group Settings

- after creating the trunk, edit the trunk
- specify the calling party and called party numbers as "Canonical with plus" for both inbound and outbound calls



## Trunk - Rights

- Make sure that you specify "No call restrictions" in the calling right, if you want the jtel System to be able to make calls to users or destinations outside of the PBX

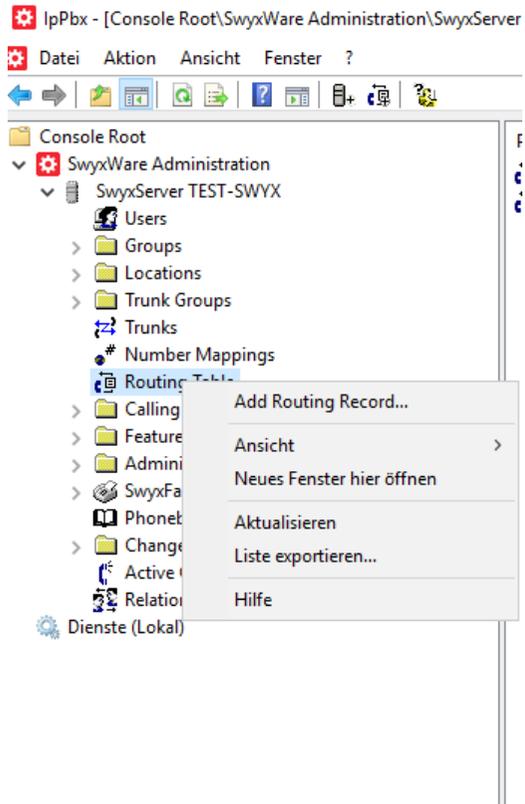


## Third step: Modify the routing table

This is where you define which calls should be routed to the jtel ACD. In this case, we are specifying calls from an outside number, +49198112233\* (\* is a wildcard).

### Add a new routing record

- Right mouse click



## Trunk Group Selection

- Specify the Trunk Group for which the rule should be applied

Add new Trunk ×

**Trunk Group Selection** 

Assign the new Trunk to an existing Trunk Group or create a new Trunk Group to which the new Trunk will be assigned.

The chosen Trunk Group determines the type of Trunk (ISDN/Analogue/SIP Gateway, SIP or ENUM Trunk, SwyxLink) and defines several common properties.

Furthermore Trunk Group settings specify if a Trunk is considered for routing outbound calls.

Trunk Group:

## Routing

- Specify a destination number - i.e. numbers which should be routed to this trunk
- Critical point: we need to specify a higher priority for the rule (e.g. 600) otherwise default routing rules in the PBX may apply.
- Here it is important, that you know your Swyx installation and how calls are routed.

Eigenschaften von 600

General Routing Source Timely Conditions

Usage

Use this Trunk Group for Calls to the following

Destination number or URI: +49198112233\*

With additional Prefix: None

Number of Retries: 0

Do not use this Trunk Group for Calls to the following

Destination number or URI:

Please enter the priority for this record. You may use a value between 0 (lowest) and 1000 (highest).

Record Priority: 600

OK Abbrechen Übernehmen Hilfe

## Fourth Step: Create a trunk group in the jtel System

### Trunk Group

- critical point: Swyx listens for incoming request on port 65002
- create a trunk group with the following configuration :



## Edit Trunk Group "Swyx"

Master Data

Trunks

ID :	6
Name :	Swyx
Incoming Number Pattern :	
SIP Source Server :	10.42.15.2:65002
SIP Destination Server :	
SIP Invited Entity :	

### Subscriber

Country Code :	49 (Germany) ▼
Area Code :	198
Subscriber Prefix :	112233

International, national and subscriber numbers are determined using these settings.

### Outgoing Trunk Selection

Trunk Group for Internal Numbers :	(Same Group) ▼
Trunk Group for External Numbers :	(Same Group) ▼
General access for outbound calls :	<input checked="" type="checkbox"/>

### Number translator

Incoming Caller :	E.164 with + incoming
Incoming Called :	E.164 with + incoming

Incoming numbers must be converted from the representation used by the signalling protocol on this trunk group to the E.164 format as used by all numbers in the portal.

Outgoing Caller : VOIP - Add prefix "sip:+", then E.164 number and postfix "@<Converter Parameter>"

Converter Parameter : 10.42.13.82

Outgoing Called : VOIP - Add prefix "sip:+", then E.164 number and postfix "@<Converter Parameter>"

Converter Parameter : 10.42.15.2:65002

Outgoing numbers must be converted from E.164 used by the portal to the format required by the signalling protocol used by this trunk group.

Outgoing send P-Asserted-Identity :

Loopback Prevention :

When the system makes an outgoing call to an agent, SIP history information (SIP history header) is provided if this is setup in the ACD group. If the history information is subsequently detected on an incoming call, then a loop has been produced (by call diversion) in the PBX. The system will reject such calls if this flag is set.

Outside Line Prefix :

Internal Number Length :

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## Trunk

- Assign the trunk for the relevant machine ID to this trunk.
- If it does not exist, create it first.

jtel PORTAL TEST2 sysadmin Change Logout DE EN

Trunks

Filter:  One Record. Showing 1 Record from 1 to 1. Page 1 of 1.

MachineID	Controller	Name	Action
82	1	test8s2	Delete

## For debugging purposes

The usual call logging applies from the jtel side. It may be necessary to run a wireshark trace to see if calls are being routed to the jtel system at all.

You can access the swyx trace logs via: <C:\ProgramData\Swyx\Traces> on the Swyx machine which may give insight as to why things are not working.

