



SETTING FOR AVAYA IPO AND DATATAL FLEXI

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1. License

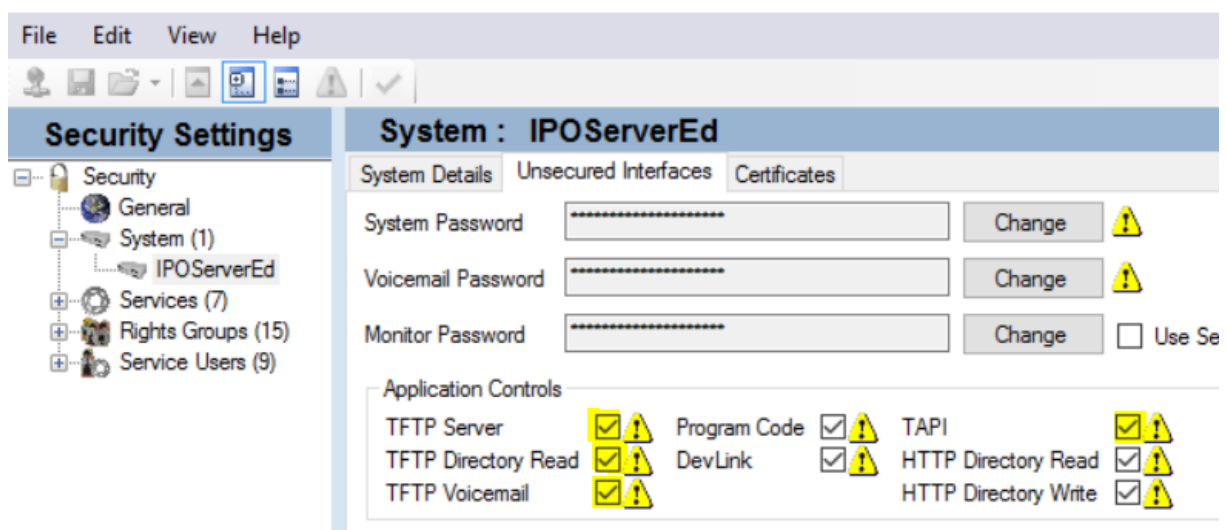
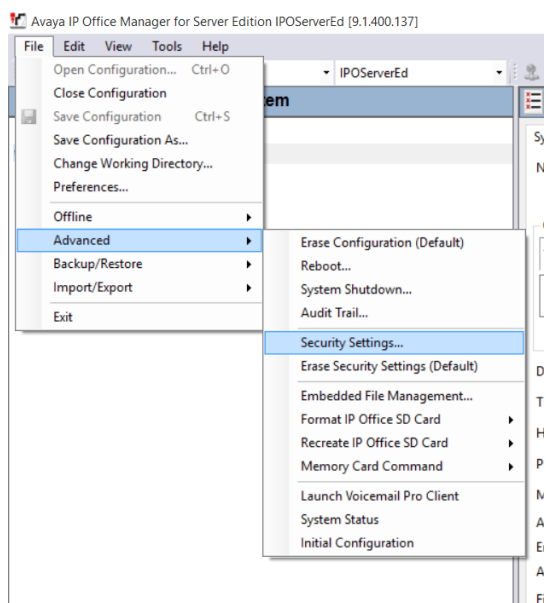
CTI Link Pro license is required

SIP Trunk Channels, 1 per channel for flexi

Flexi is tested and verified for Avaya IP Office 9.x and higher, lower version of Avaya IP Office is not supported by Datatal.

2. Security settings

In Manager for Avaya IP Office, go to Security settings and activate TAPI and TFTP



3. SIP trunk

Add a new SIP Line

SIP Line

ITSP domain Name: <PBX IP>

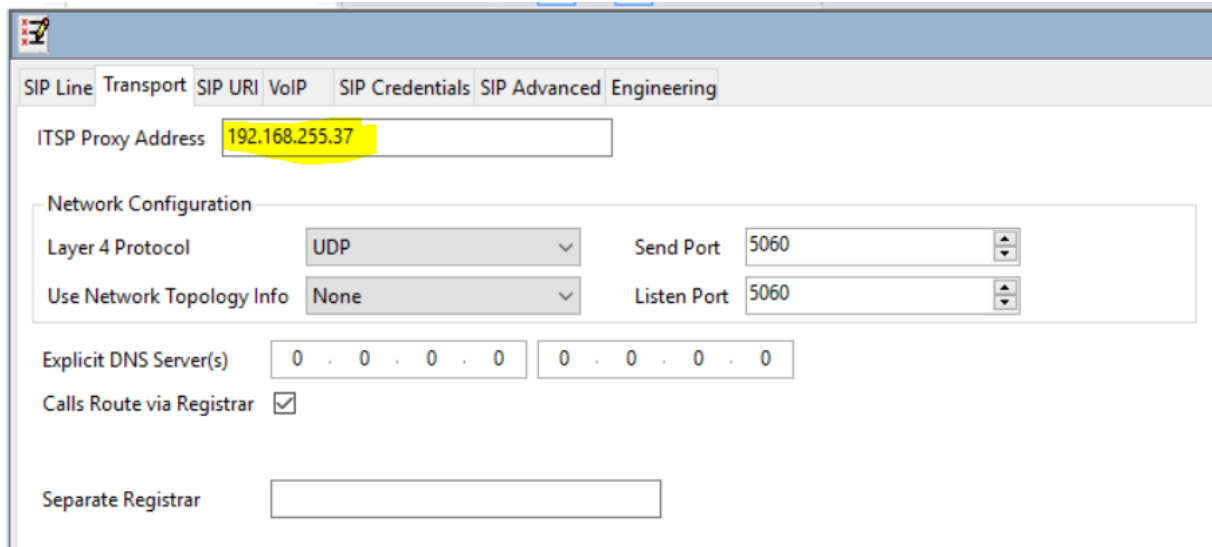
Send Caller ID: P-Asserted ID

REFER Support: Always (both incoming and outgoing)

		SIP Line - Line 9	
<div style="display: flex; justify-content: space-between;"> SIP Line Transport SIP URI VoIP SIP Credentials SIP Advanced Engineering </div>			
Line Number	9	In Service	<input checked="" type="checkbox"/>
ITSP Domain Name	192.168.255.70	Check OOS	<input type="checkbox"/>
URI Type	SIP	Session Timers	
Location	Cloud	Refresh Method	Reinvite
Prefix		Timer (seconds)	On Demand
National Prefix	0	Forwarding and Twinning	
International Prefix	00	Originator number	
Country Code		Send Caller ID	P Asserted ID
Name Priority	System Default	Redirect and Transfer	
Description		Incoming Supervised REFER	Always
		Outgoing Supervised REFER	Always
		Send 302 Moved Temporarily	<input type="checkbox"/>
		Outgoing Blind REFER	<input type="checkbox"/>

Transport

ITSP Proxy Address: <Flexi Server IP>



The screenshot shows a configuration window with a blue header bar containing a logo. Below the header is a tabbed interface with the following tabs: SIP Line, Transport (selected), SIP URI, VoIP, SIP Credentials, SIP Advanced, and Engineering. The main content area is divided into several sections:

- ITSP Proxy Address:** A text input field containing the IP address 192.168.255.37, which is highlighted in yellow.
- Network Configuration:** A section containing:
 - Layer 4 Protocol:** A dropdown menu set to UDP.
 - Send Port:** A numeric input field set to 5060.
 - Use Network Topology Info:** A dropdown menu set to None.
 - Listen Port:** A numeric input field set to 5060.
- Explicit DNS Server(s):** Two IP address input fields, both containing 0 . 0 . 0 . 0.
- Calls Route via Registrar:** A checkbox that is checked.
- Separate Registrar:** An empty text input field.

SIP URI

Add a SIP URI

Outgoing Group: <Set a number; make a note of it, it will be needed later>

Channel	Groups	Via	Local URI	Contact	Display Name	PAI	Credential	Max Calls
1	1	<... *	*	*		N...	0: <Non...	11

Edit Channel

Via: <None>

Local URI:

Contact:

Display Name:

PAI:

Registration: 0: <None>

Incoming Group:

Outgoing Group:

Max Calls per Channel:

VoIP

Compression Mode: G.711 ALAW 64K

SIP Line	Transport	SIP URI	VoIP	SIP Credentials	SIP Advanced	Engineering
SIP Line - L						
Codec Selection		System Default		<input checked="" type="checkbox"/> Re-invite Supported <input checked="" type="checkbox"/> Codec Lockdown <input type="checkbox"/> Allow Direct Media Path <input type="checkbox"/> Force direct media with phones <input type="checkbox"/> PRACK/100rel Supported <input type="checkbox"/> G.711 Fax ECAN		
Unused		Selected				
G.711 ULAW 64K		G.722 64K G.729(a) 8K CS-ACELP G.711 ALAW 64K				
Fax Transport Support		None				
DTMF Support		RFC2833/RFC4733				
Media Security		Disabled				

SIP Advanced, no changes needed

SIP Line - Line 9*

SIP Line Transport SIP URI VoIP SIP Credentials SIP Advanced Engineering

Addressing

Association Method **By Source IP address**

Call Routing Method **Request URI**

Suppress DNS SRV Lookups

Identity

Use Phone Context

Add user=phone

Use + for International

Use PAI for Privacy

Use Domain for PAI

Swap From and PAI

Caller ID from From header

Send From In Clear

Cache Auth Credentials

User-Agent and Server Headers

Media

Allow Empty INVITE

Send Empty re-INVITE

Allow To Tag Change

P-Early-Media Support **None**

Send SilenceSupp=Off

Force Early Direct Media

Media Connection Preservation **Disabled**

Call Control

Call Initiation Timeout (s) **4**

Call Queuing Timeout (m) **5**

Service Busy Response **486 - Busy Here**

on No User Responding Send **408-Request Timeout**

Action on CAC Location Limit **Reject Call**

Suppress Q.850 Reason Header

Emulate NOTIFY for REFER

No REFER if using Diversion

Shortcode

Add a new shortcode

Code: 95xxxx (x:s = extension length)

Feature: Dial

Telephone Number: Nss

Line Group ID: <Enter same number that was chosen in SIP URI>

Short Code

Code

Feature

Telephone Number

Line Group ID

Locale

Force Account Code

Incomming Call route

Add a new Incomming Call route for Flexi trunk

Destination is a dot

IP Office	Line Group ID	Incoming Number	Destination
<Shared>	1	.	.

Presently installations, later in Flexi setup you will be asked to enter divert destination, enter "95[EXT]" except quotation marks

User – No answer and busy

Tab -> Forwarding

Forward Number: <Shortcode+extension>

Forward On Busy

Forward On No Answer

Forward Number

Forward Internal calls

Twinned user and TAPI

Add this value under user NoUser

The screenshot shows the Flexi Admin web interface for user 'NoUser'. The 'Source Numbers' tab is selected, and the 'Source Number' field contains the value 'TAPI_REPORTS_TWIN_CALLS'.

Setup FNE service

For dialing via Flexi Presentity App it's require a FNE service setup in PBX. FNE feature will be call from mobile phone and Flexi will send DTMF tones PBX some call will be redirected to destination.

This feature can be turned off for specific users in Flexi AdmiTal web, users->premissions->users accessgroup->user-> [setting] Call through company

Create a shortcode with FNE 31

The screenshot shows the Flexi Admin web interface for creating a new shortcode. The 'Code' field is '*75', the 'Feature' is 'FNE Service', and the 'Telephone Number' is '31'. A red warning message states: '* This Short Code is common to all systems.' The 'Line Group ID' is '0' and the 'Locale' is set to a default value. The 'Force Account Code' checkbox is unchecked.

Add an incoming call route to this FNE service. Remember the Incoming Number, will be require for setup guide.

Line Group ID is from network carrier trunk

☰
20 0498253033

Standard
Voice Recording
Destinations

* This Incoming Call Route is common to all systems.

Bearer Capability	Any Voice
Line Group ID	20
Incoming Number	0498253033
Incoming Sub Address	
Incoming CLI	
Locale	
Priority	1 - Low
Tag	
Hold Music Source	System Source
Ring Tone Override	None

☰
20 0498253033

Standard
Voice Recording
Destinations

	TimeProfile	Destination	Fallback Extensio
▶	Default Value	*75	▼

Diversion using shortcode

Create a shortcode as below. *23*<diversionCode># will be the command on phone to initiate a diversion.

In admital web, login as sysop goto system->admin->settings and tab HVD and FunctionInterceptCode (Hänvisningskommandokod) should be set to 23.

Short Code

Code

Feature

Telephone Number

Line Group ID

Locale

Force Account Code

OK Cancel

Disable Inhibit Off-Switch Forward/Transfer

This is require, otherwise forwarded calls to Flexi SIP trunk won't work.

When enabled, this setting stops any user from transferring or forwarding calls externally. See Off-Switch Transfer Restrictions.

Maximum SIP sessions

This field is shown for Server Edition systems. On Server Edition systems, the Maximum SIP Sessions value must match the total number of SIP set and trunk calls that can occur at the same time.

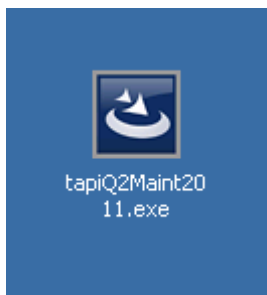
The Maximum SIP Sessions setting determines the number of SIP Trunk Channel licenses reserved for concurrent sessions on any SIP trunks provided by the server. Those licenses are reserved from the pool of SIP Trunk Channel licenses in the configuration of the Primary Server.

The screenshot shows the configuration interface for IPOServerEd, specifically the Telephony settings. The 'Maximum SIP Sessions' field is set to 50 and is highlighted in yellow. The 'Inhibit Off-Switch Forward/Transfer' checkbox is checked and highlighted in yellow. Other settings include Dial Delay Time (1), Dial Delay Count (30), Default No Answer Time (29), Hold Timeout (25), Park Timeout (300), Ring Delay (5), Call Priority Promotion Time (Disabled), Default Currency (SEK), Default Name Priority (Favour Trunk), Media Connection Preservation (Enabled), and Phone Failback (Manual). The 'Companding Law' section shows 'A-Law' selected for both Switch and Line. Other options like 'DSS Status', 'Auto Hold', 'Dial By Name', 'Show Account Code', 'Restrict Network Interconnect', 'Drop External Only Impromptu Conference', 'Visually Differentiate External Call', 'High Quality Conferencing', and 'Directory Overrides Barring' are also visible.

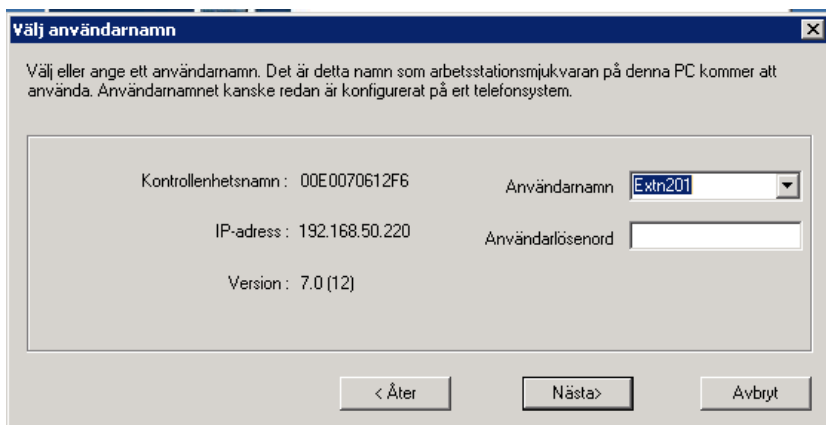
NOTE! Next step TAPI installation

4. TAPI install

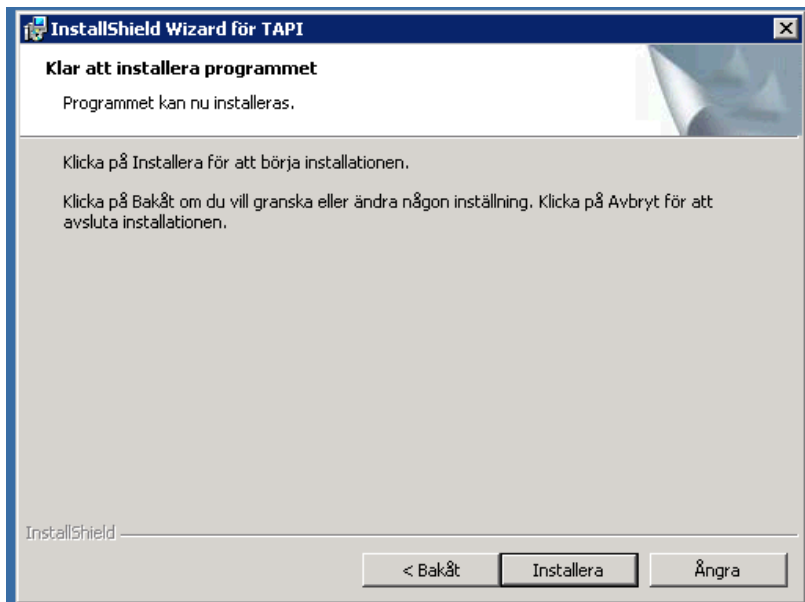
From Avaya IPO UserCD, install TAPI-service provider (TSP) copy it to desktop



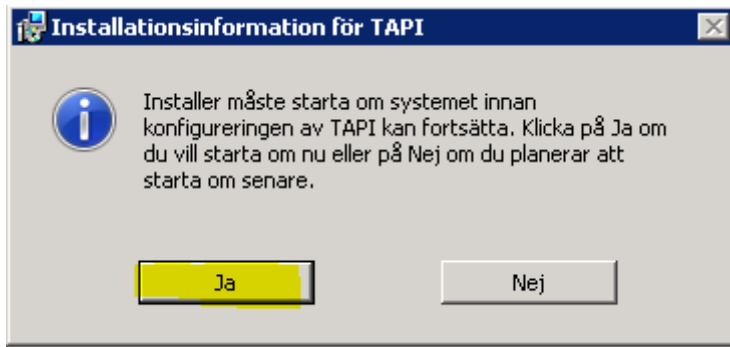
Start installation and proceed to this message box popup. Just press **next**,



Press **install**

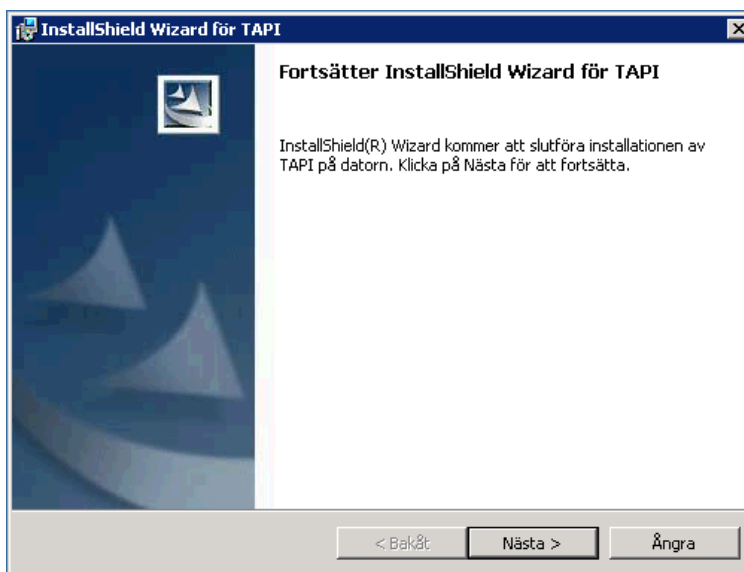


Press **YES** at next popup

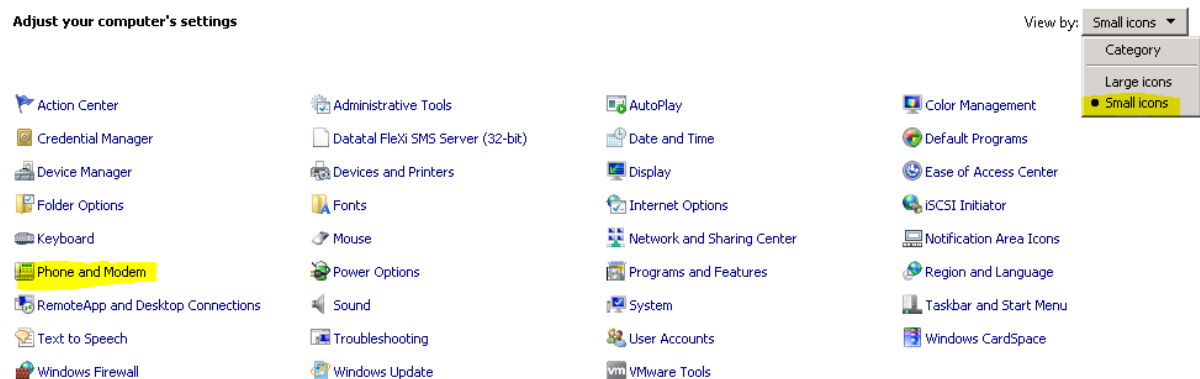


After reboot...

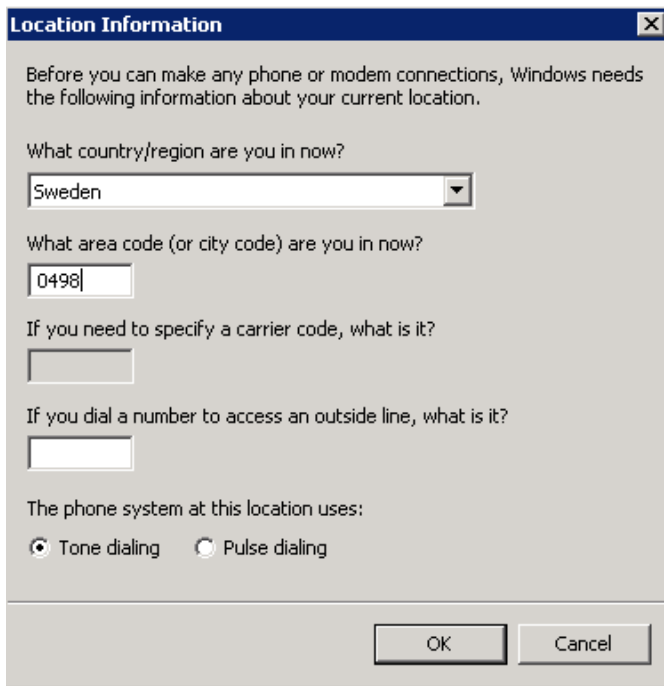
Finish Avaya TAPI setup



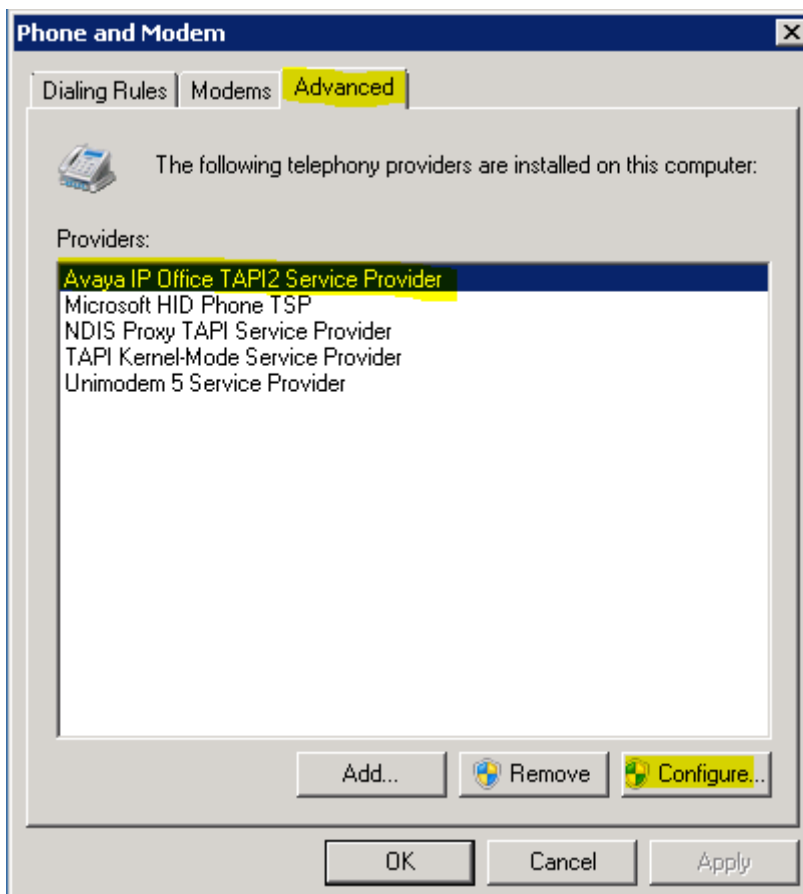
Go to control panel and change view to small icons, and start "Phone and Modem"



Enter area code, and OK



Select tab **Advanced**, make sure that “**Avaya IP office TAPI2 Service Provider**” is selected and press **Configure..**



Switch IP Address: Enter PBX IP

Select “Third Party”

Switch Password: This is set in PBX, default is 'password'

Select "ACD QUEUES"

Avaya TAPI2 configuration

Switch IP Address 192.168.255.70

OK

Cancel

Single User

User Name

User Password

Third Party

Switch Password

Ex Directory Users

WAV Users

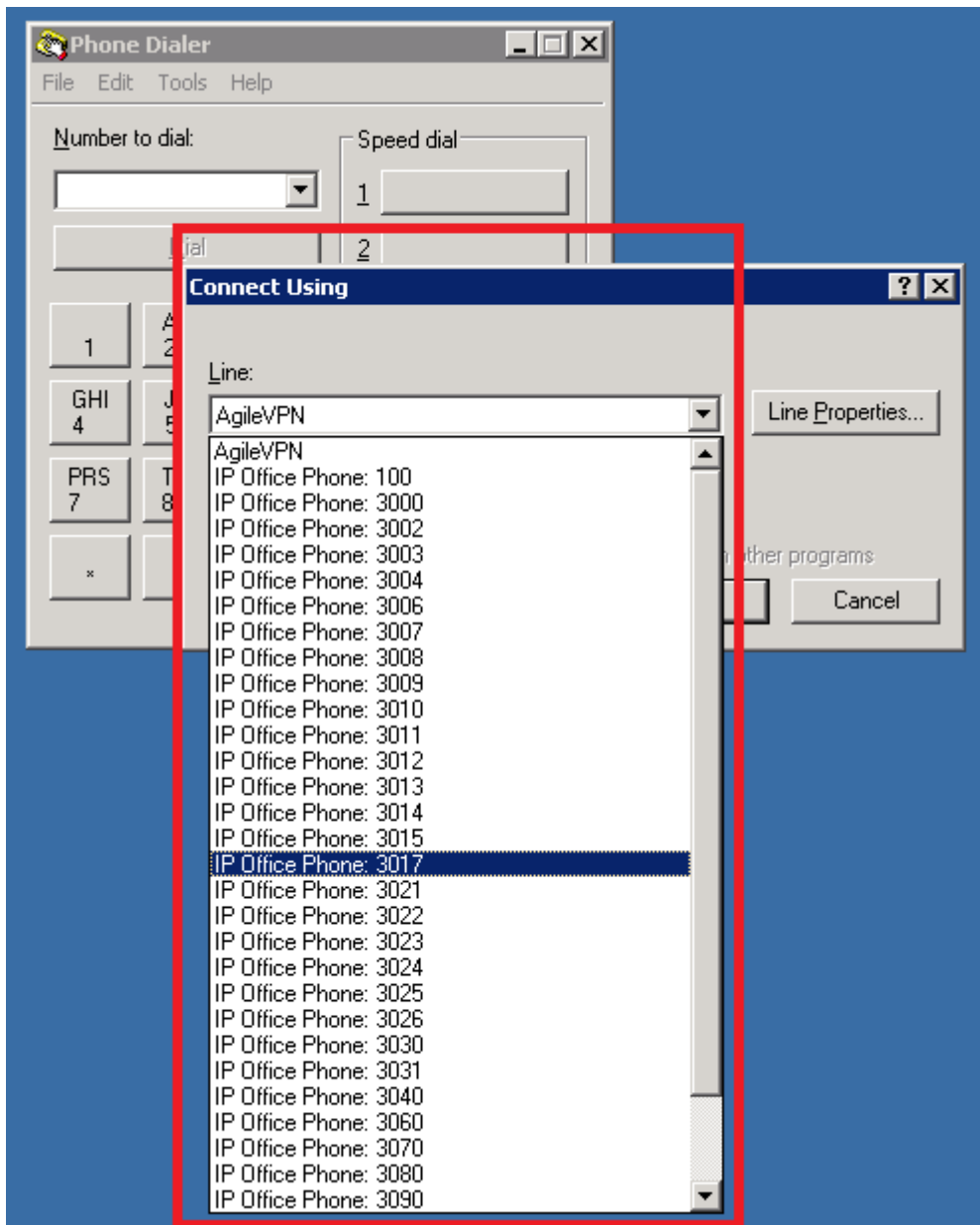
ACD Queues

Press OK and reboot server

After reboot, test connection. Press **start** and **run..**

Enter **dialer** under line, you should now see your extension in PBX

if not check password in PBX to "Avaya TAPI2 configuration"



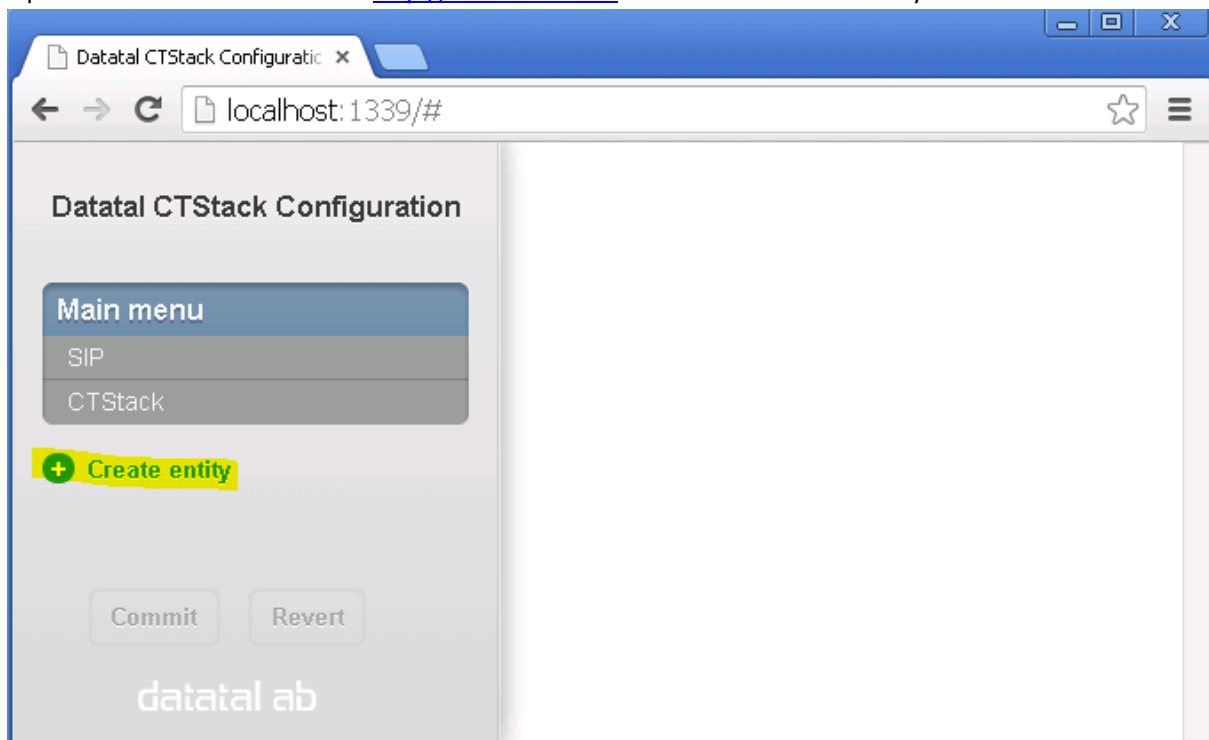
Next, install Flexi Server from FlexiInstaller interface

5. Datatal CTStack settings

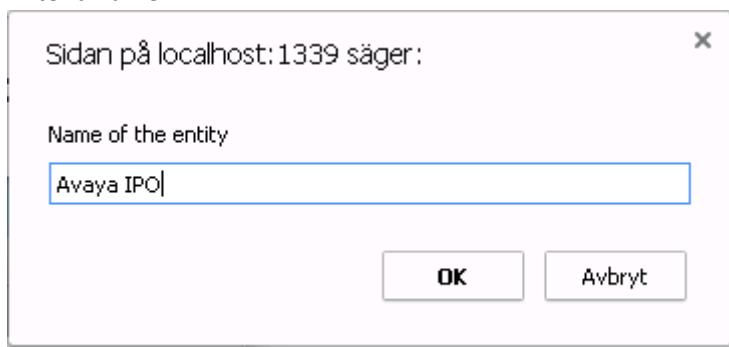
Require: Internet explorer 10 or higher, Chrome or Firefox, websockets support

If upgrade for < Flexi 5.9, Dialogic Diva will be replaced with Datatal CTStack and configuration is imported from Dialogic diva softIP

1. Open a browser and browse to <http://localhost:1339> and click on "Create entity"



2. Enter a name



- Click on SIP
 - Park other calls on Makecall: uncheck
 - Play "ring" at other calls on Makecall: checked

localhost:1339

Datatall CTStack Configuration

Main menu

- CTStack
- SIP

flexi-ipo.redneck.se x

- Media
- SIP
- Telephony

+ Create entity

2 change(s) pending

Commit
Revert

datatal ab

flexi-ipo.redneck.se - SIP

Dialogs

Dialogs

- Always create early dialogs: ?
- Retry-After 4xx: ?
- Use OPTIONS for keep-alive: ?

Outbound

- Always use proxy: ?
- Outbound proxy: ?
- Set 'Diversion' header on MakeCall: ?
- Set 'History-Info' header on MakeCall: ?

Transfer

- Park other calls on MakeCall: ?
- Play 'ring' at other calls on MakeCall: ?
- Terminate local call transfer on INVITE: ?
- Treat BYE as transfer success: ?
- Use 'Remote-Target' in 'Refer-To': ?
- Wait for park complete on MakeCall: ?

Scroll down...

Check "Use 'From' header"

localhost:1339

Datatal CTStack Configuration

Main menu

- CTStack
- SIP

flexi-ipo.redneck.se

- Media
- SIP
- Telephony

+ Create entity

3 change(s) pending

Commit **Revert**

datatal ab

Use 'Remote-Target' in 'Refer-To':

Wait for park complete on MakeCall:

Registrations

Users

Registrations:

ADD
EDIT
REMOVE

SIP Dialogs

Use 'From' header:

RFC 3325

P-*-Identity mode:

Use P-Asserted-Identity:

Transport

Transport:

Commit changes

3 change(s) pending

Commit **Revert**

4. Click on Telephony

Address: Customers main number, like 0498253000

Name: Enter a namn

Default: SIP URI host: PBX-IP address

Trunk mode: Check

Lines: Enter number of lines that is order, if entered 20 lines and the license is valid for 16 lines, CTstack will only use 16 lines simultaneous

Datatal CTStack Configuration

Main menu

- CTStack
- SIP

flexi-ipo.redneck.se ✕

- Media
- SIP
- Telephony

+ Create entity

5 change(s) pending

Commit **Revert**

datatal ab

flexi-ipo.redneck.se - Telephony

Line configuration

Standard

BlindCall source mode:

INVITE expires:

Lines:

SIP

Address

Address:

Default domain:

Default SIP URI host:

Default SIP URI port:

Name:

Profile

Apply:

Current:

Trunk

Trunk mode:

5. Commit and commit changes

5 change(s) pending

Commit **Revert**

datatal ab

No changes need under media

Datatal CTStack Configuration

Main menu

- CTStack
- SIP

flexi-ipo.redneck.se ✕

- Media
- SIP
- Telephony

+ Create entity

flexi-ipo.redneck.se - Media

Codec

RTP

Default RTP codec:

RTP

Audio

Send silent RTP frames:

Networking

Max RTP port:

Min RTP port:

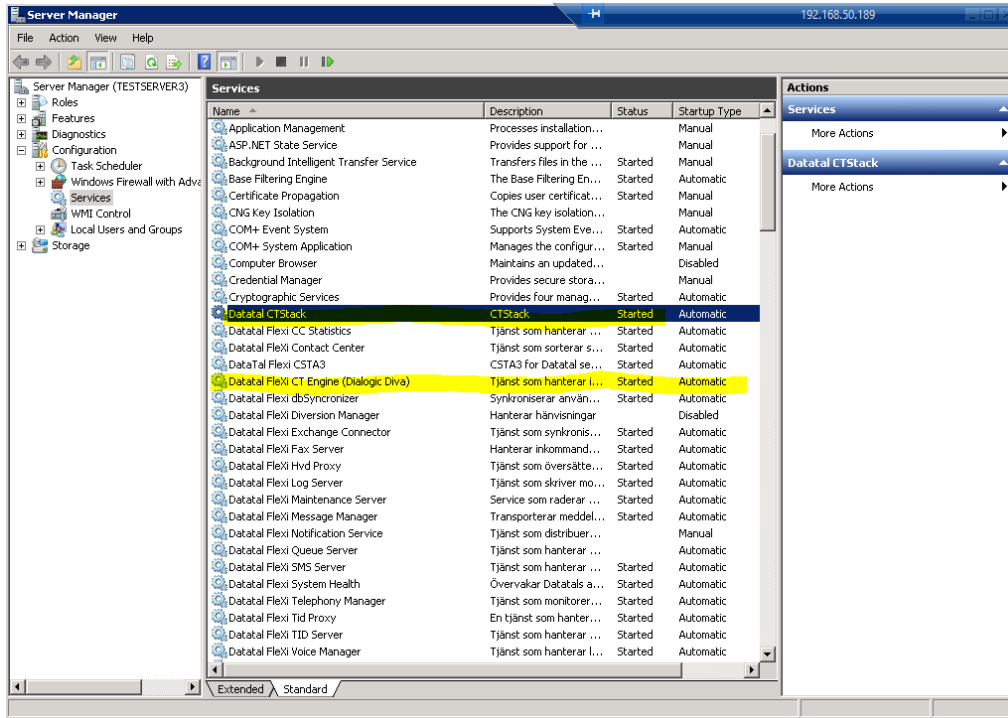
SDP

Media

OnHold attribute:

ptime:

STOP and START CTStack and CTENGINE services, when number of lines are change these two service has to be restarted



6. Configure multiple accessnumber for call though pbx

Create Accessnumber

Require Flexi server 5.11.1

Log in to Admital web. To enable accessnumber configuration, goto user->premission->[select premission group]->accessnumber. Set List, Edit, Create and delete. Then save,

The screenshot shows the Admital web interface at www.datatal.se. The breadcrumb navigation is **User / Permissions / Edit access group**. On the left, a menu lists options: Index, Messages, User (highlighted), Permissions, Edit access group (with a dropdown arrow), Save, and Copy hot keys. The main content area shows a list of properties: Properties, Accessnumber (highlighted), Admin, Alarm, Blind, Button controlled menu, Call Center, Company, and Company greetings. To the right, under the heading **Accessnumber**, there are four checked checkboxes: List, Edit, Create, and Delete.

New option will appear, click on it

Accessnumber->New accessnumber (company)

Name: Display name for accessnumber

Telephony number: Full DDI number, recommended to use with country code

Country code: Country code for accessnumber, this will be used if user intend to call national numbers without country code.

Accessnumber

The screenshot shows the 'Accessnumber' configuration form. The form has a blue header with the title 'Accessnumber'. Below the header, there are four input fields: 'Company' with the value 'Datatal AB', 'Name' with the value 'Norway', 'Telephone number' with the value '+47498253017', and 'Country code' with the value '47'. At the bottom of the form, there are three buttons: 'Save', 'Cancel', and 'Apply'.

System

On system level you can configure a accessnumber that will be default for all that has not been override.

Login into Admital as sysop user, goto system->admin->settings->telephony and MEX address is system wide setting

www.datatal.se [System](#) / [Admin](#) / [Settings](#)

[Index](#)
[Messages](#)
[User](#)
[Call Center](#)
[Company](#)
[Interception](#)
[Queue/transfer](#)
[Greetings](#)
[Menu](#)
[Voice mail](#)
[Schedule](#)
[Entry point](#)
[Statistics](#)
[Search](#)
[Time booking](#)
[System](#)
[Admin](#)
[Settings](#)
[Monitor](#)
[Nodes](#)

Telephony

Country code: +46
Area code: 0498
Number prefix: 25
Extension length: 4
MEX address: +46498253033
Outdial no answer timeout: 25
Anonymous addresses: anonymous
restricted

Company

On company level you can configure the default, can be override on user level

Log in as sysop user and goto company and select the company you intend to configure.

[Company](#) / [Edit company](#)

Company

Name: Datatal AB

Permit diversion to different destination

Accessnumber: [+46498253033 (System)]

User

On user level, goto user and select user, under tab phone and select accessnumber. This setting is override all other accessnumber settings

The screenshot shows the web interface for managing users on the website www.datatal.se. The page title is "User / Edit user". On the left, a navigation menu includes "Index", "Messages", "User", "Permissions", "Save", "Copy hot keys", "Call Center", and "Company". The "User" menu item is highlighted in yellow, and its sub-item "Edit user" is also highlighted in yellow. The main content area is titled "Phone" and contains a form with the following fields: "Accessnumber:" (highlighted in yellow) with a dropdown menu showing "Datatal Företag Test", "Mobile:", "Fax nr:", "Alt. tele:", and "PSTN:". A vertical sidebar on the left of the main content area lists navigation options: "User", "Logon", "Data", "Phone" (highlighted in yellow), "Links", "Voice mail", and "Image".