

# Status Presence, analog devices and OpenSIP in Teams Direct Routing Era



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## Comunicazione vocale

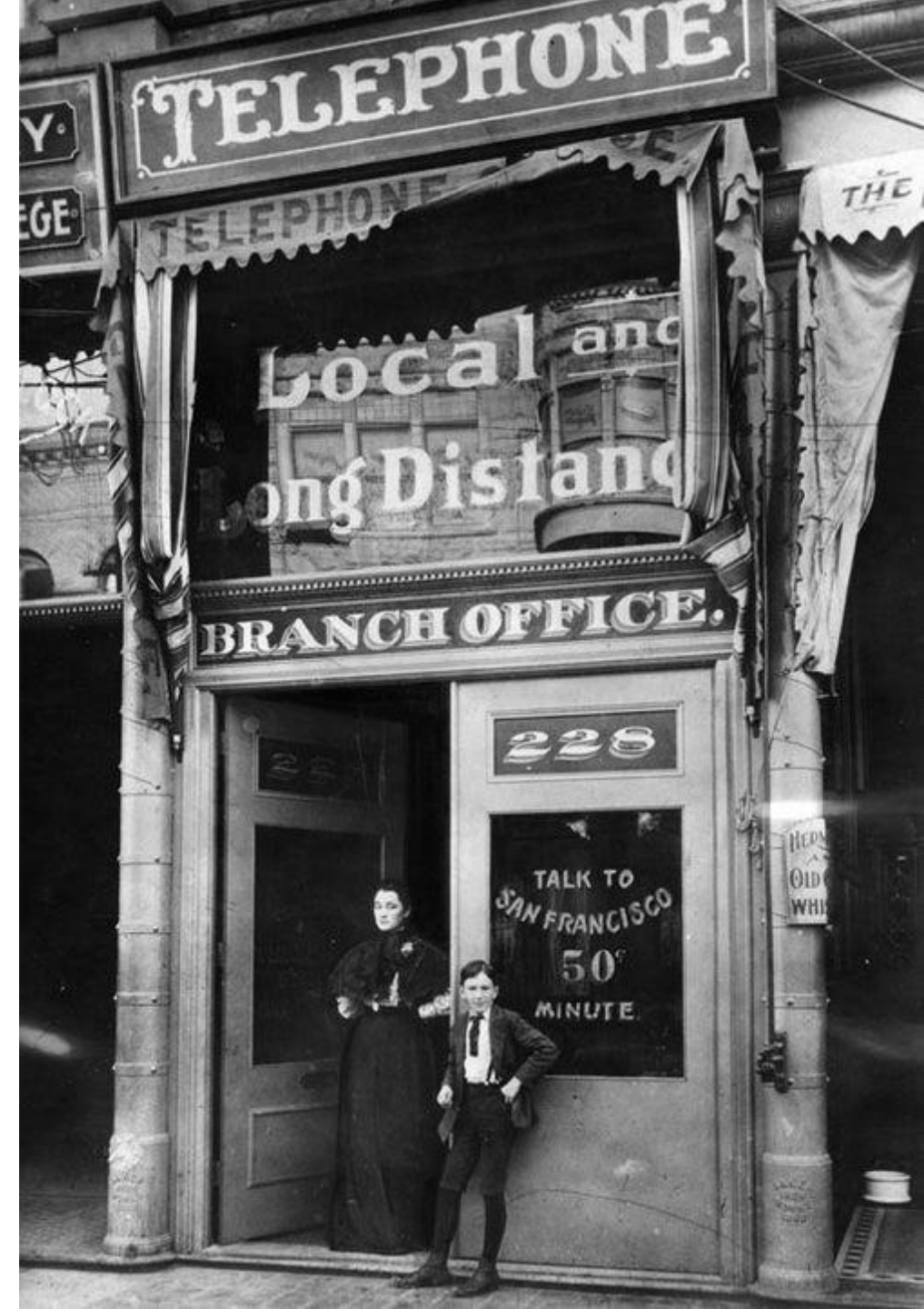
la voce è lo strumento più importante con il quale comunichiamo

## Adozione

l'utilizzo degli impianti Skype for Business e Teams integrati con la fonia passa dal 10-15% al 90% (dati CC)

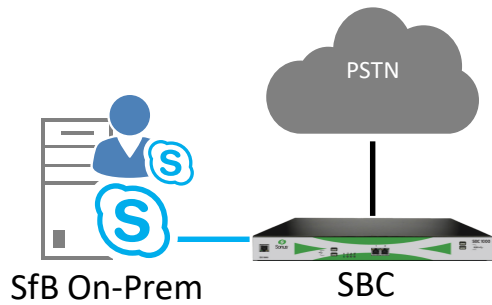
## Complessità

l'Enterprise Voice è l'argomento più complesso e delicato legato a Skype for Business e Teams

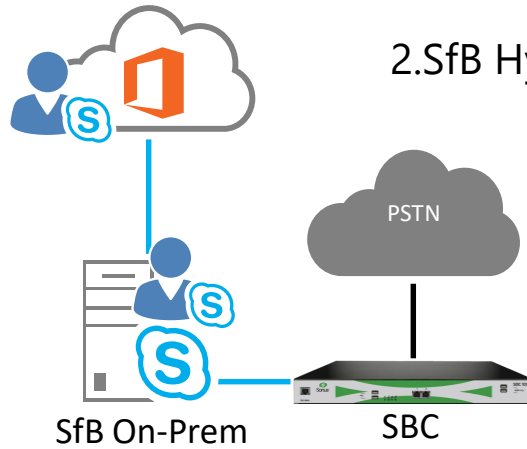


Skype for Business e Microsoft Teams: scenari di fonia

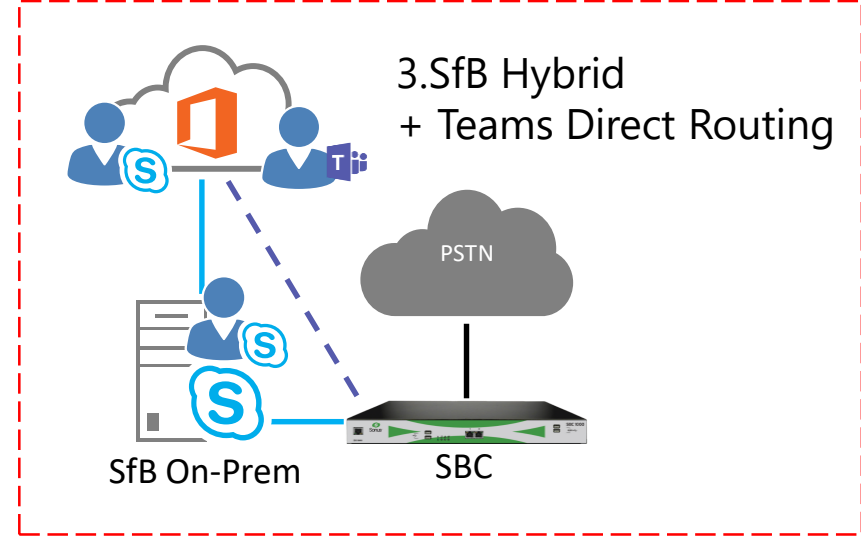
1.SfB On-Prem



2.SfB Hybrid



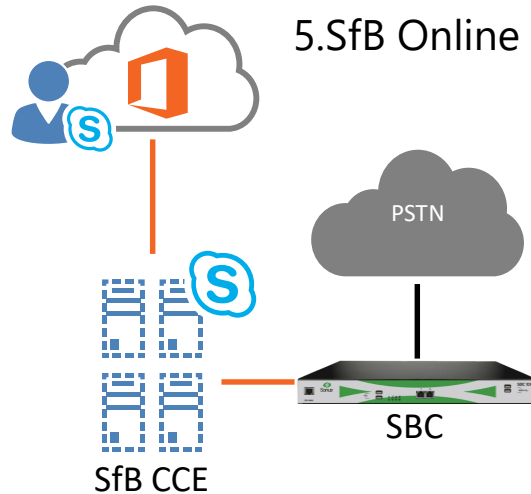
3.SfB Hybrid  
+ Teams Direct Routing



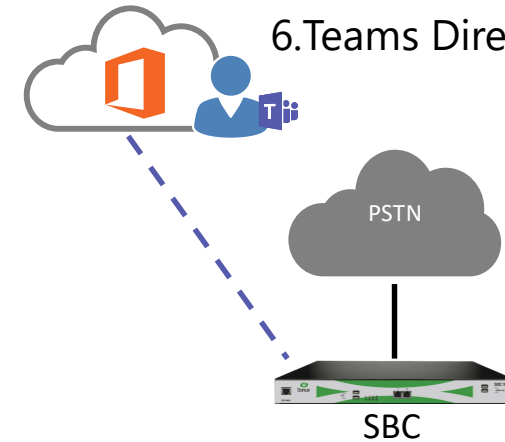
4.SfB Online / Teams  
+ Microsoft Calling Plan

NON DISPONIBILE IN ITALIA  
Disponibile in AU, BE, CA, FR,  
DE, IE, PR, NL, ES, UK, USA

5.SfB Online + CCE



6.Teams Direct Routing



## Upstream e Downstream

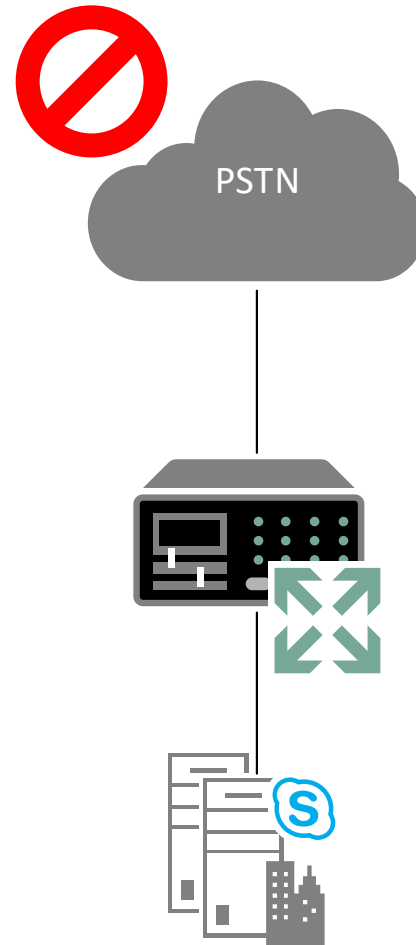
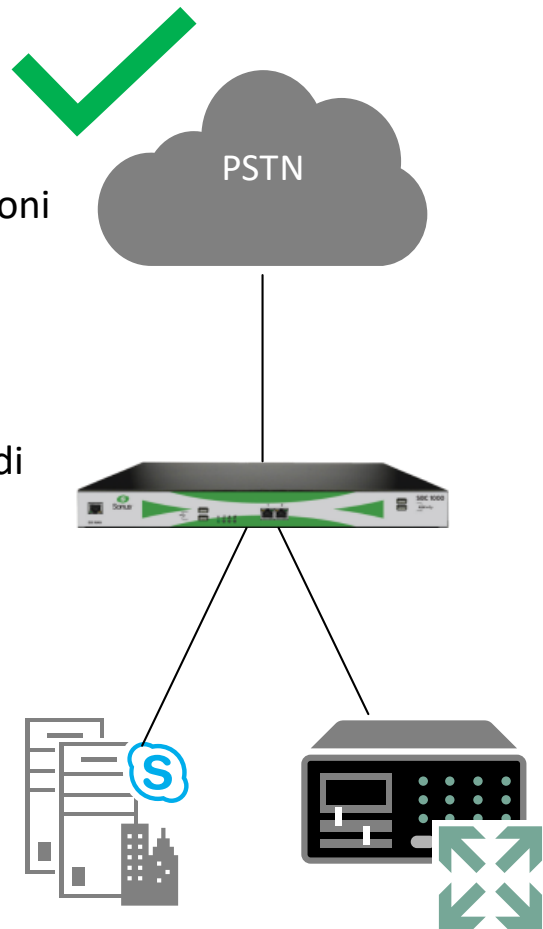
## Upstream

## PRO

Configurazione ideale per le migrazioni  
Consente il controllo completo della  
fonia

## CONTRO

Richiede una conoscenza completa di  
tutti gli aspetti legati alla fonia e del  
Voice Gateway utilizzato



## Downstream

## PRO

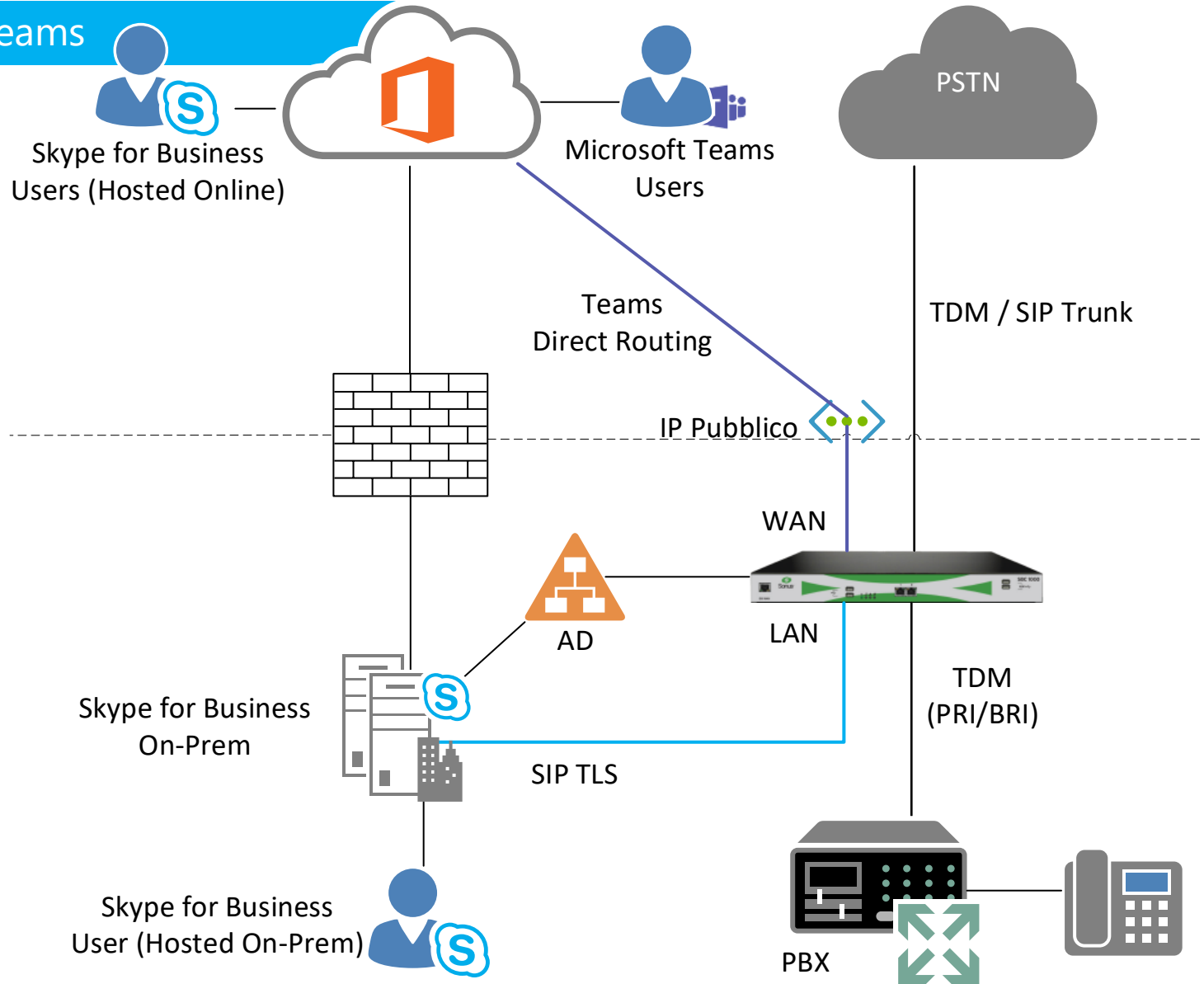
Non si "intacca" la fonia esistente  
Veloce da implementare per i POC

## CONTRO

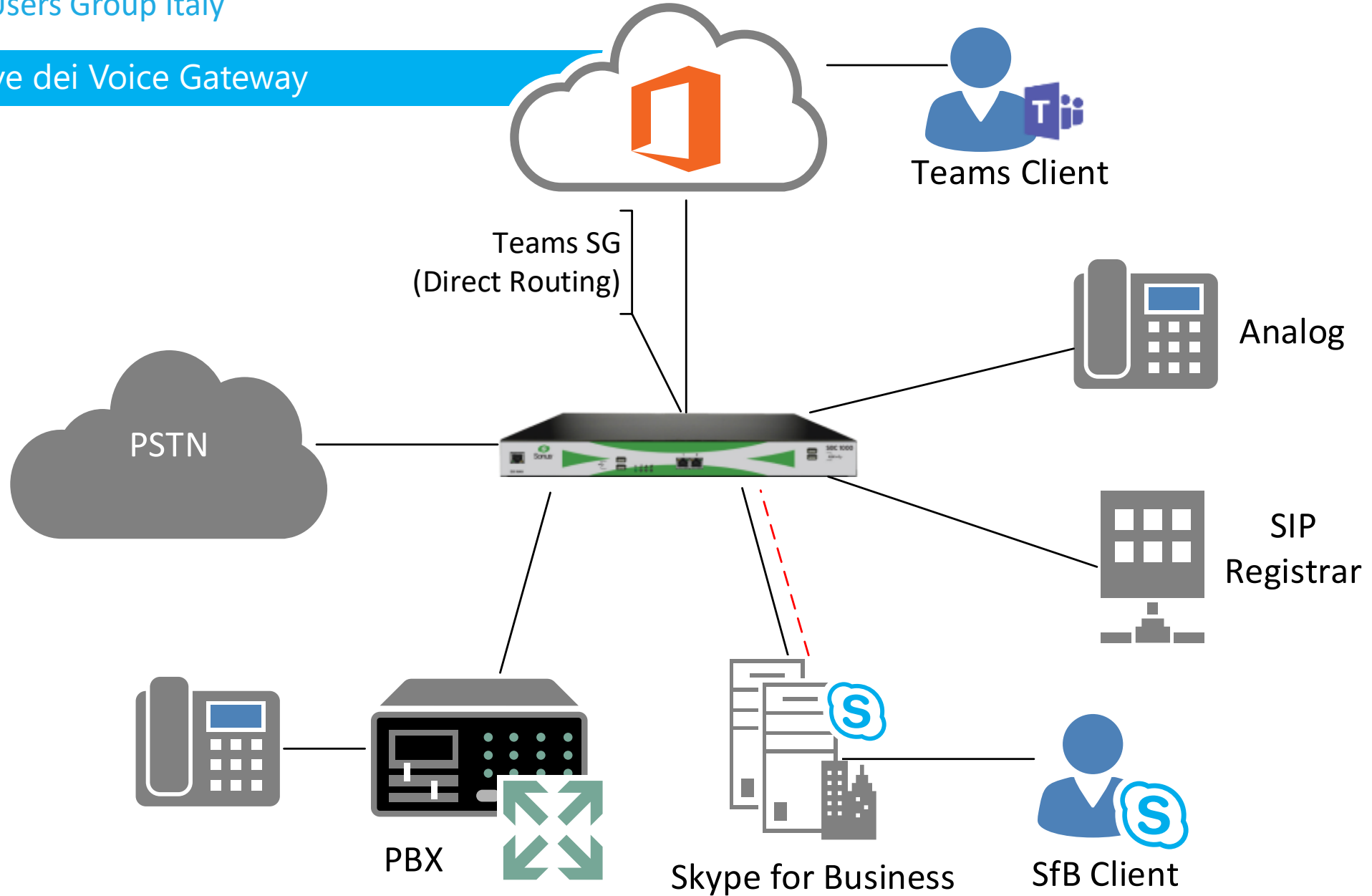
Assenza totale di controllo sulla fonia  
Si è legati a scelte e configurazioni  
decise da chi gestisce il PBX esistente,  
che spesso ignorano e a volte  
ostacolano l'introduzione di soluzioni  
di UC come SfB e Teams

Nessuna migrazione seria alla UC parte  
da questa configurazione

# Esempio di deployment SfB e Teams



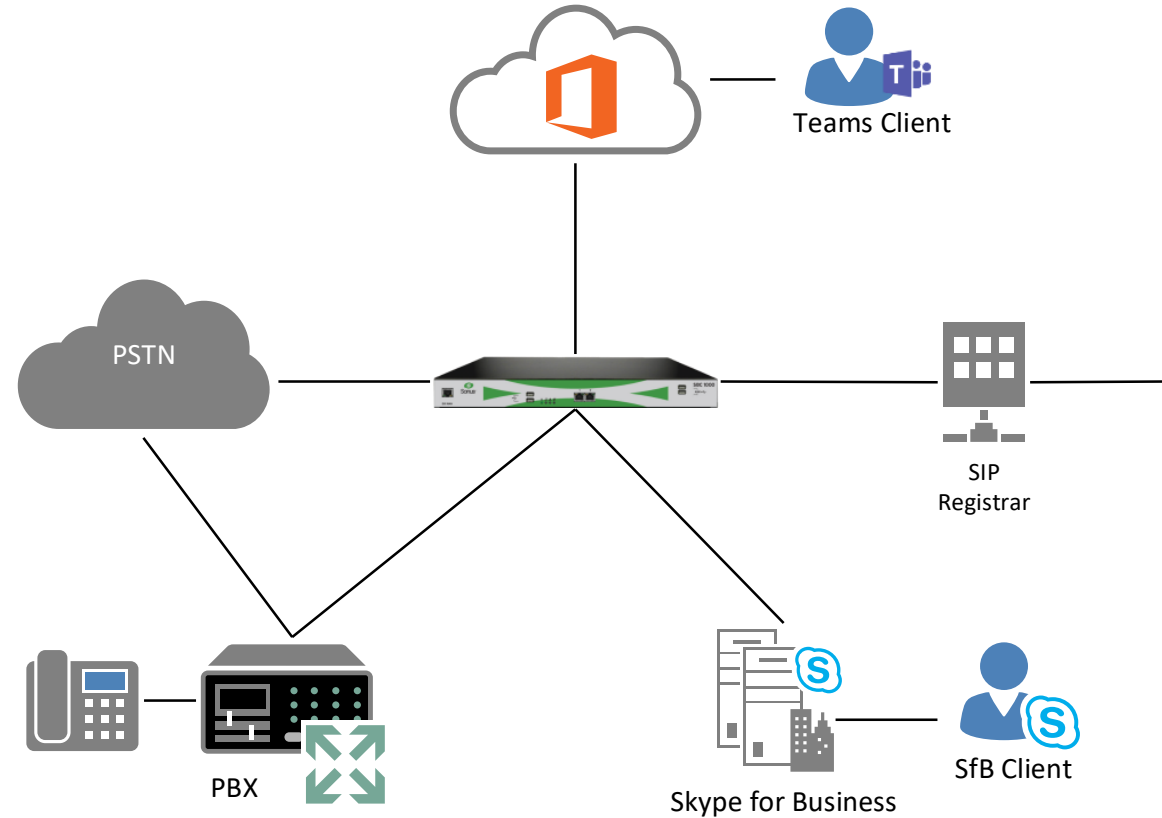
Il ruolo chiave dei Voice Gateway



## SIP Registrar

Risponde a una frequente domanda dei clienti:  
E' possibile riutilizzare i telefoni dell'attuale PBX  
basato su Standard SIP (non MS) con  
Skype for Business e Teams?

Probabilmente si, accettando alcune limitazioni



# SIP Registrar: Setup

ribbon Monitor Tasks Settings Diagnostics System Welcome: Itvitali | Logout  
 Device Name: voicegw Ribbon SBC

September 25, 2018 00:47:03

### Signaling Group Table

Total 6 Signaling Group Rows

Type	Description	Admin State	Service Status	Display	Primary Key
SIP	UC	Up	Up	Counters   Channels   Sessions	1
SIP	TEAMS	Up	Up	Counters   Channels   Sessions	2
SIP	REGISTRAR	Up	Up	Counters   Channels   Sessions	3

Description: REGISTRAR  
 Admin State: Enabled  
 Service Status: Up

### SIP Channels and Routing

Action Set Table: None  
 Call Routing Table: REGISTRAR  
 No. of Channels: 50  
 SIP Profile: Default SIP Profile  
 SIP Mode: Local Registrar  
 Registrar: Local Registrar  
 Agent Type: Back-to-Back User Agent  
 Registrar Min. TTL: 600  
 Load Balancing: Round Robin  
 Channel Hunting: Standard  
 Notify Lync CAC Profile: Disable  
 Challenge Request: Disable  
 Outbound Proxy:   
 Outbound Proxy Port: 5060  
 No Channel Available Override: 34: No Circuit/Channel Available  
 Call Setup Response Timer: 255  
 Call Proceeding Timer: 180

### Media Information

Audio/Fax Stream Mode: DSP Proxy Direct  
 Video/Application Stream Mode: Disabled  
 Media List ID: SIP Media List  
 Play Ringback: Auto on 180/183  
 Tone Table: Italy Tone Table  
 Play Congestion Tone: Disable  
 Early 183: Disable  
 Allow Refresh SDP: Enable  
 Music on Hold: Disabled

### Mapping Tables

SIP To Q.850 Override Table: Default (RFC4497)  
 Q.850 To SIP Override Table: Default (RFC4497)  
 Pass-thru Peer SIP Response Code: Enable

### Listen Ports

Total 2 SIP Listen Port Rows

Port	Protocol	TLS Profile ID
5060	UDP	N/A
5060	TCP	N/A

### SIP IP Details

Signaling/Media Source IP: Ethernet 1 IP (10.1.100.155)  
 Signaling DSCP: 40  
 Static NAT - Outbound:   
 Outbound NAT Traversal: None  
 Static NAT - Inbound:   
 Detection: Disabled

### Federated IP/FQDN

Total 2 SIP Federated IP Rows

IP/FQDN	Netmask/Prefix
10.0.0.0	255.0.0.0
192.168.0.0	255.255.0.0

### SIP Local Registrar Table

Total 1 SIP Local Registrar Row

Description	Max. Users	Display
Local Registrar	50	Counters   Registered U

Description: Local Registrar  
 Maximum Number of Users: 50

Apply



# SIP Registrar: Analog Accounts

SIP Registrar User Table: Local Registrar - Internet Explorer

SIP Registrar User Table: Local Registrar September 23, 2018 00:27:11

Clear Users Total 2 SIP Registrar User Rows

<input type="checkbox"/>	User Name	Source IP	Source Port	Public Source IP	Public Source Port	Transport	Last Updated	Time Until Expiration
<input type="checkbox"/>	807	10.1.61.19	5060	N/A	N/A	UDP	2018/09/23 - 00:02:37	00:35:27
<input type="checkbox"/>	805	192.168.80.105	5060	N/A	N/A	UDP	2018/09/23 - 00:09:54	00:42:44

Test Siemens Contact Properties

Member Of: General | Object: Address | Security: Telephones

Test Siemens Contact

First name:  Initials:

Last name:

Display name:

Description:

Office:

Telephone number:

E-mail:

Web page:

OK Cancel Apply Help

```

Identity : CN=Test Siemens Contact
VoicePolicy : Test Account
VoiceRoutingPolicy :
RegistrarPool : ccsfbfe01.
Gateway : voicegwcento.
AnalogFax : False
Enabled : True
SipAddress : sip:testsiemens.contact@
LineURI : tel:+3905 805
DisplayName : Test Siemens Contact
DisplayNumber : +3905 805
ExUmEnabled : False
  
```

```

Identity : CN=Test WiFi Contact,
VoicePolicy : Test Account
VoiceRoutingPolicy :
RegistrarPool : ccsfbfe01.
Gateway : voicegwcento.
AnalogFax : False
Enabled : True
SipAddress : sip:testwifi.contact@
LineURI : tel:+3905 807
DisplayName : Test WiFi Contact
DisplayNumber : +3905 807
ExUmEnabled : False
  
```

Test WiFi Contact Properties

Member Of: General | Object: Address | Security: Telephones | Attribute Editor: Organization

Test WiFi Contact

First name:  Initials:

Last name:

Display name:

Description:

Office:

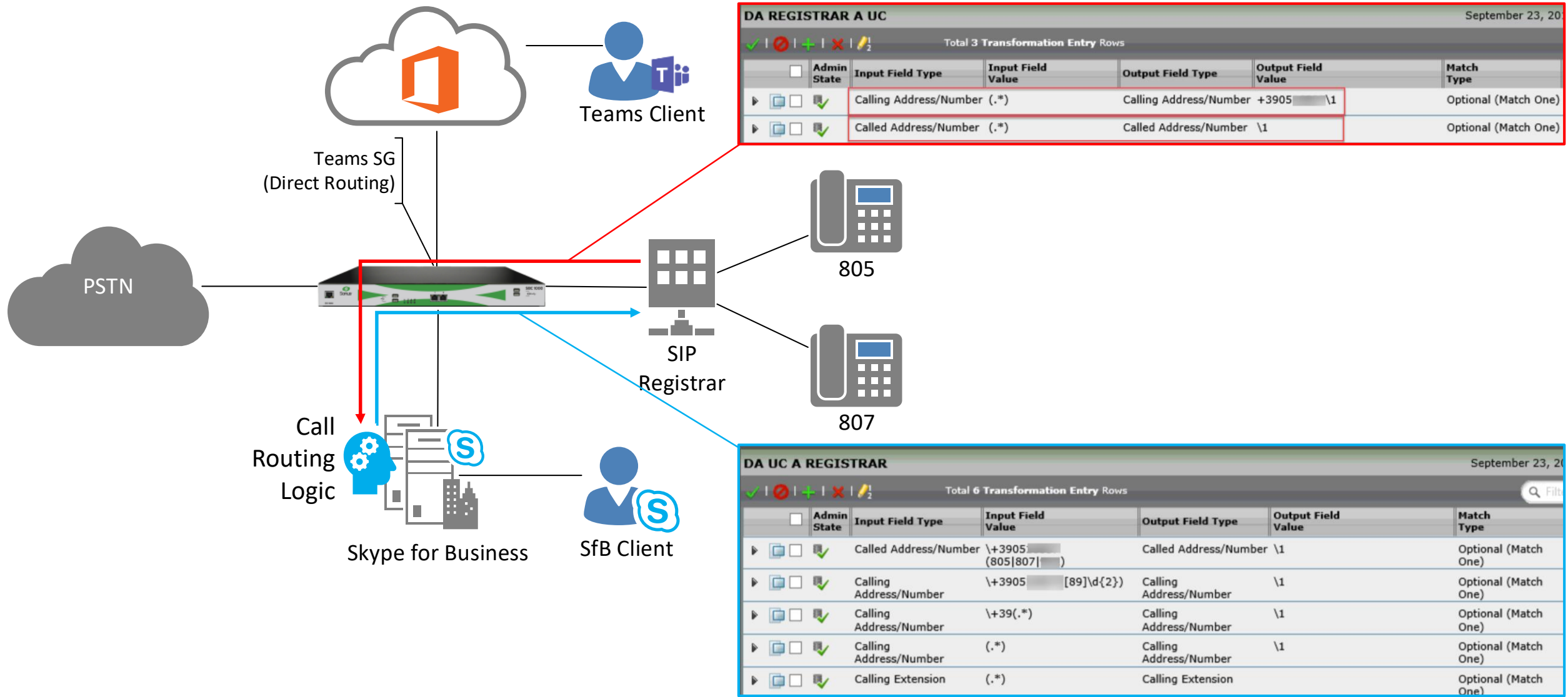
Telephone number:

E-mail:

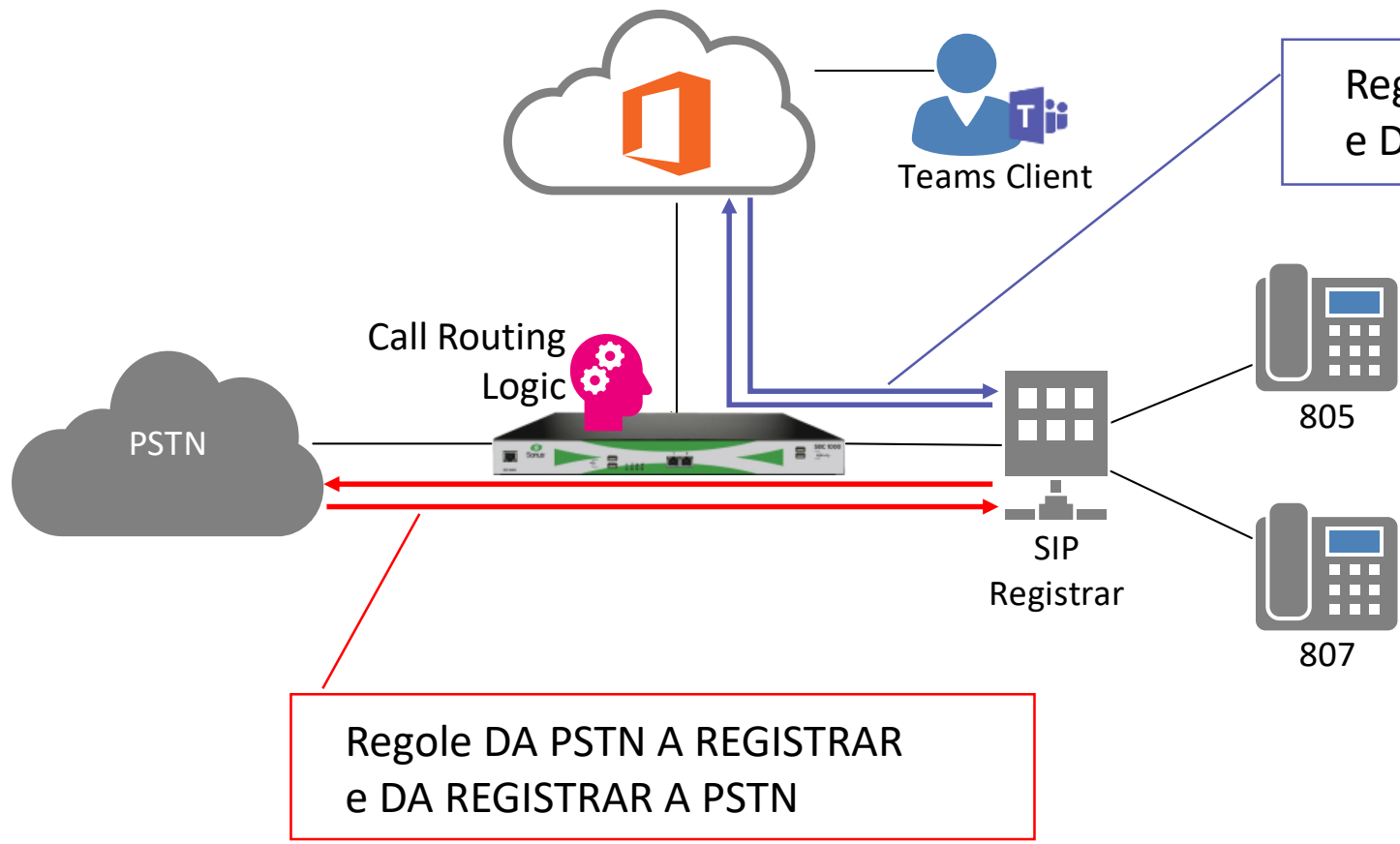
Web page:

OK Cancel Apply Help

SIP Registrar: Call Routing via SfB



SIP Registrar: Call Routing w/o SfB



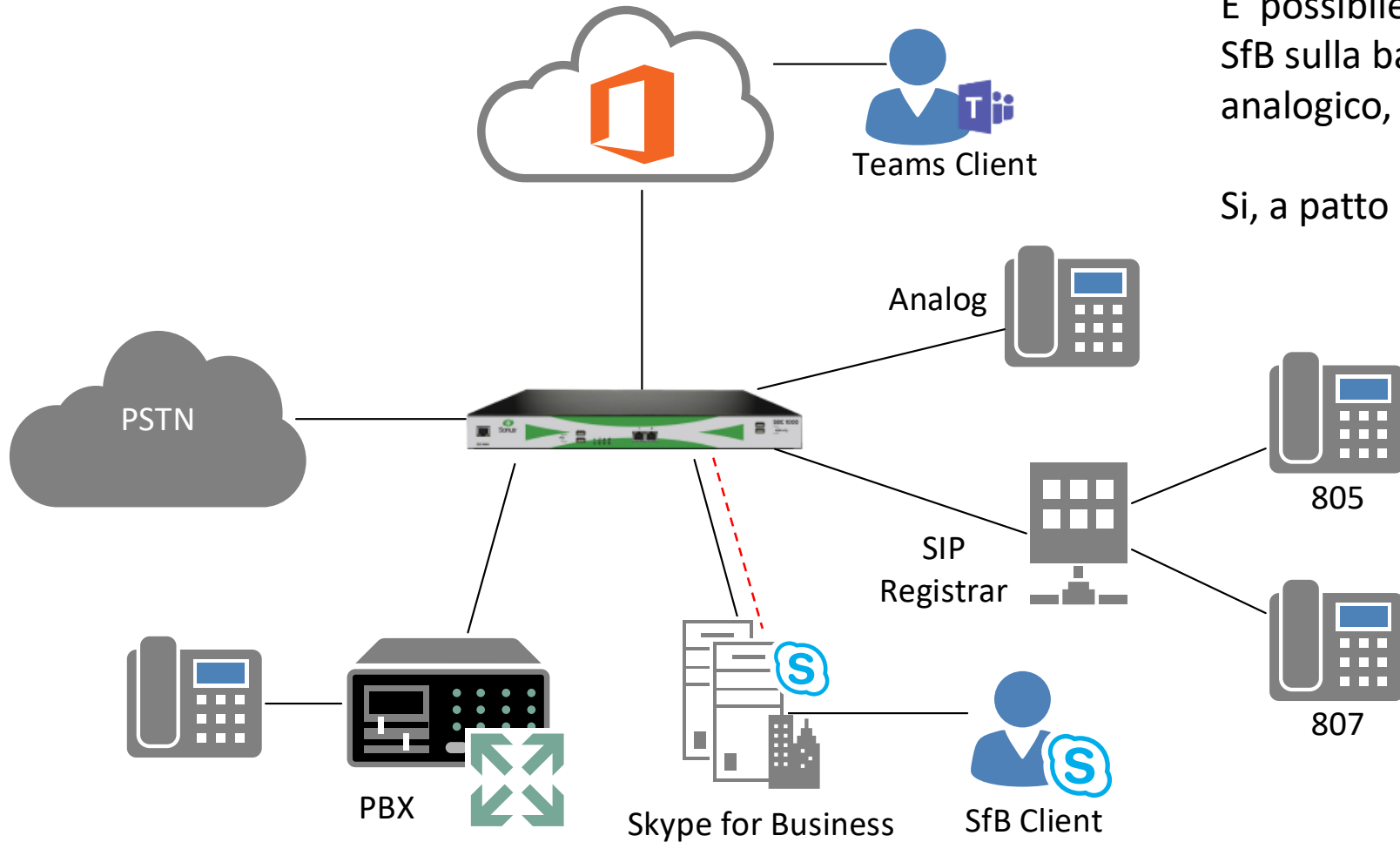
Regole DA TEAMS A REGISTRAR  
e DA REGISTRAR A TEAMS

Regole DA PSTN A REGISTRAR  
e DA REGISTRAR A PSTN

A differenza di SfB On-Prem, Microsoft Phone System (su cui si basa Teams per la fonia) non supporta nativamente i device analogici, non abbiamo quindi le funzioni di Voice Policy

La logica di Call Routing è spostata tutta sui Voice Gateway, su cui devono risiedere tutte le regole di instradamento e i DialPlan

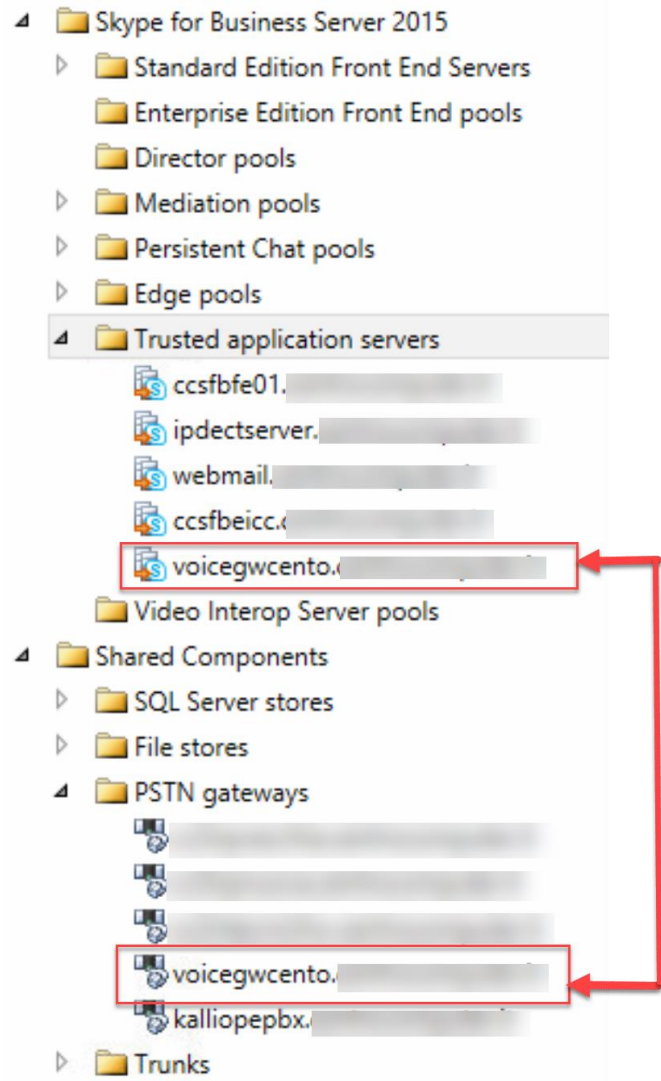
Status Presence



Risponde a una domanda dei clienti:  
E' possibile modificare lo stato di presenza di un utente SfB sulla base di una chiamata in corso con un device analogico, OpenSIP o collegato a un PBX esistente?

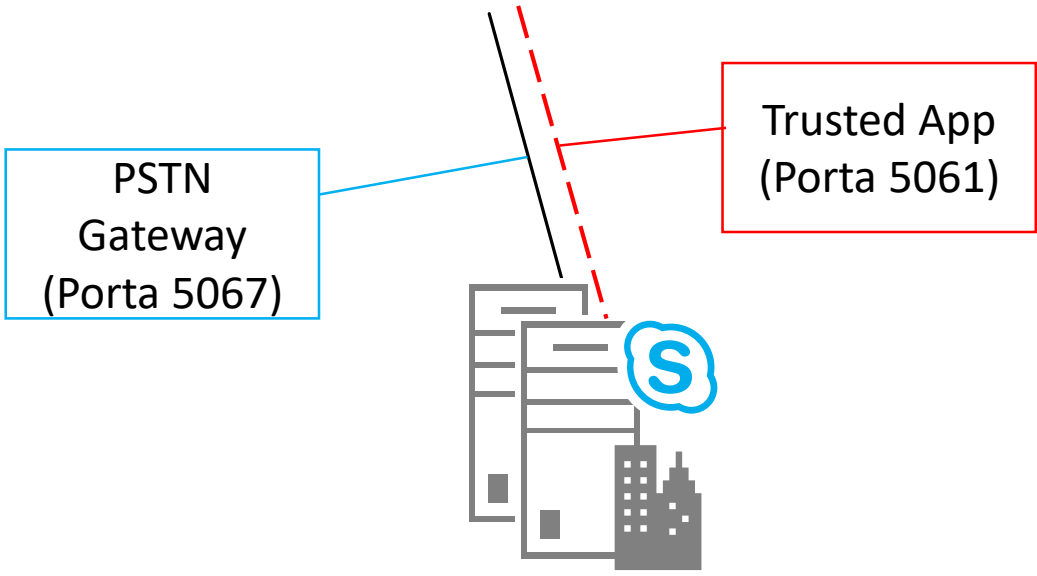
Si, a patto che la chiamata transiti attraverso l'SBC

# Trusted App Setup



```

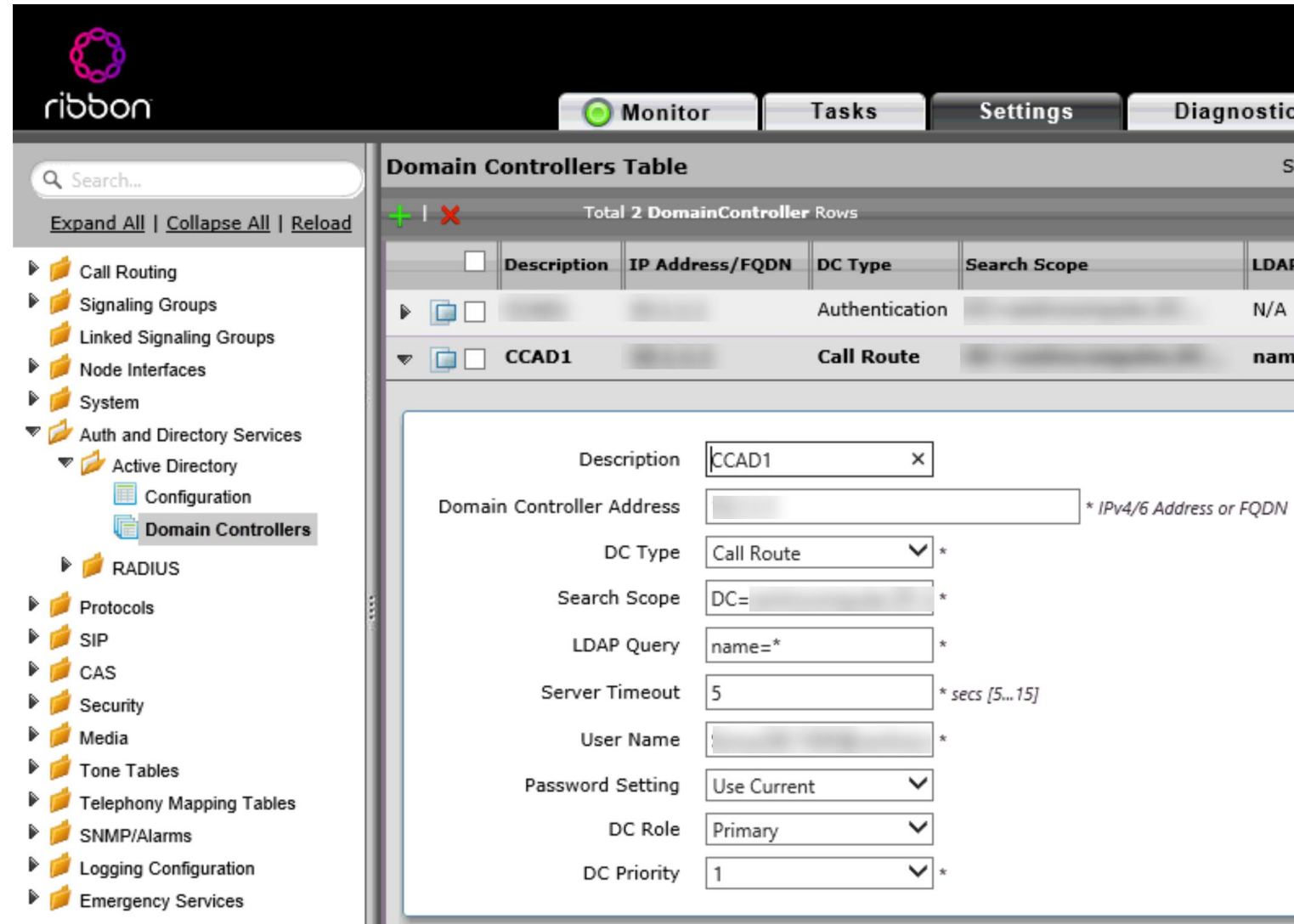
New-CsTrustedApplicationPool -Identity <SBC FQDN> -Registrar <FE POOL FQDN> -Site 1
New-CsTrustedApplication -ApplicationId "sonuspresence" -TrustedApplicationPoolFqdn <SBC FQDN> -Port 5061
Enable-CsTopology
    
```



Supportato da:  
Lync Server 2013 v5.0.8308.866  
Skype for Business Server 2015-2019

Skype for Business

AD Setup



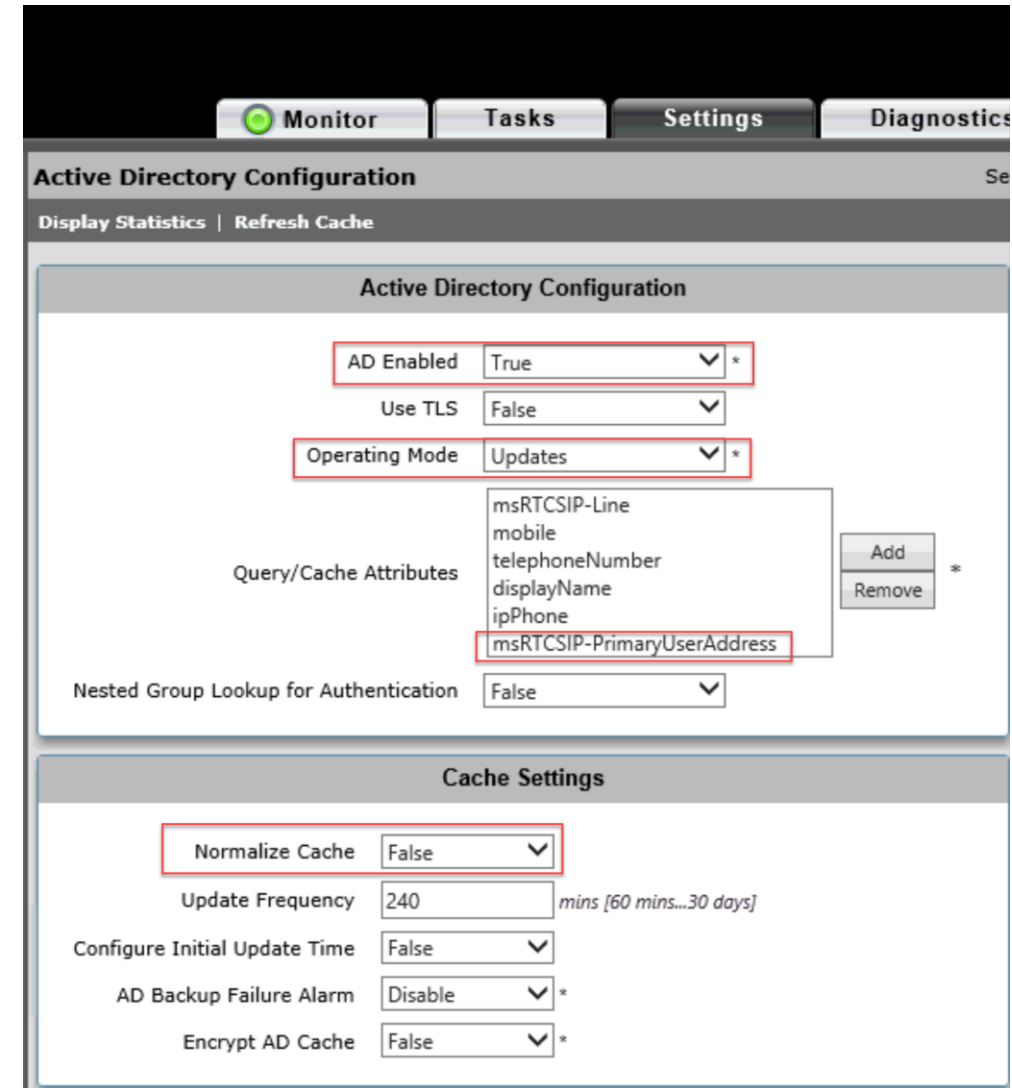
**Domain Controllers Table**

Total 2 DomainController Rows

Description	IP Address/FQDN	DC Type	Search Scope	LDAP
CCAD1		Call Route		nam

Configuration details for CCAD1:

- Description: CCAD1
- Domain Controller Address: [Redacted] \* IPv4/6 Address or FQDN
- DC Type: Call Route \*
- Search Scope: DC= [Redacted] \*
- LDAP Query: name=\* \*
- Server Timeout: 5 \* secs [5...15]
- User Name: [Redacted] \*
- Password Setting: Use Current
- DC Role: Primary
- DC Priority: 1 \*



**Active Directory Configuration**

AD Enabled: True \*

Use TLS: False

Operating Mode: Updates \*

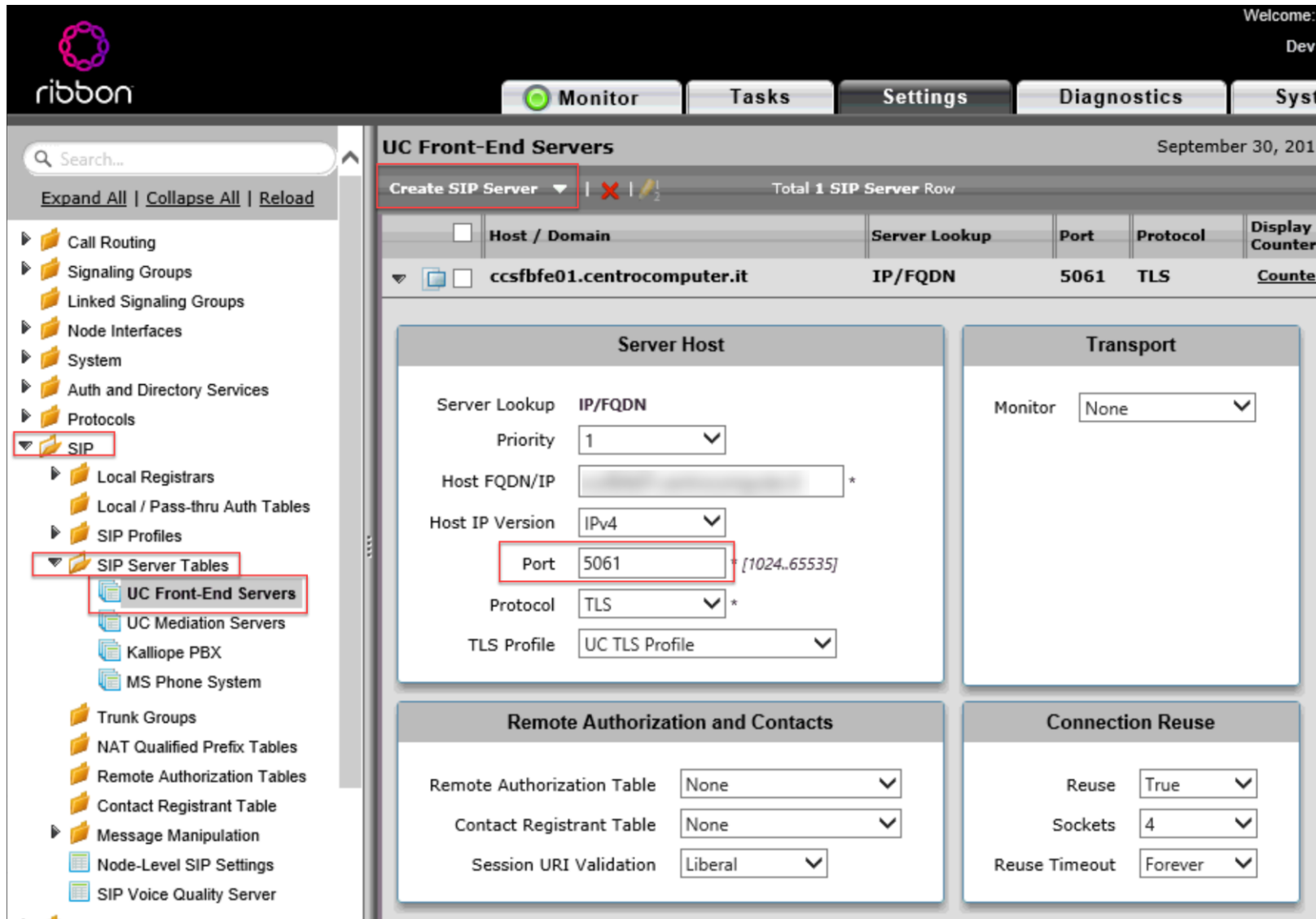
Query/Cache Attributes: msRTCSIP-Line, mobile, telephoneNumber, displayName, ipPhone, msRTCSIP-PrimaryUserAddress \*

Nested Group Lookup for Authentication: False

**Cache Settings**

- Normalize Cache: False \*
- Update Frequency: 240 mins [60 mins...30 days]
- Configure Initial Update Time: False
- AD Backup Failure Alarm: Disable \*
- Encrypt AD Cache: False \*

Presence Server



The screenshot shows the Ribbon Communications Management Console interface. The left sidebar contains a navigation tree with the following items: Call Routing, Signaling Groups, Linked Signaling Groups, Node Interfaces, System, Auth and Directory Services, Protocols, SIP (highlighted), Local Registrars, Local / Pass-thru Auth Tables, SIP Profiles, SIP Server Tables (highlighted), UC Front-End Servers (highlighted), UC Mediation Servers, Kalliope PBX, MS Phone System, Trunk Groups, NAT Qualified Prefix Tables, Remote Authorization Tables, Contact Registrant Table, Message Manipulation, Node-Level SIP Settings, and SIP Voice Quality Server.

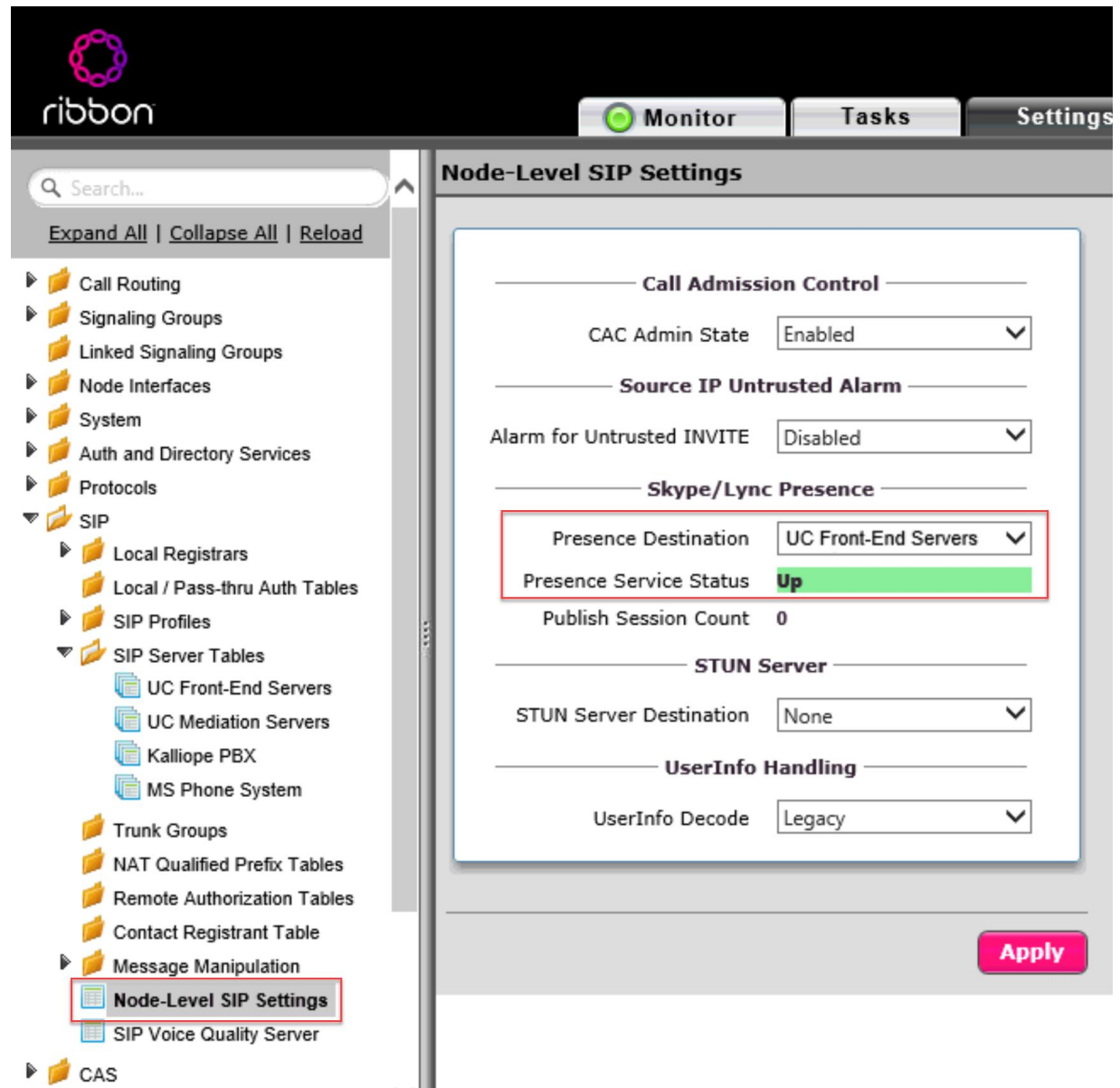
The main content area is titled "UC Front-End Servers" and shows a table with one row:

Host / Domain	Server Lookup	Port	Protocol	Display Counter
<input type="checkbox"/> ccsfbfe01.centrocomputer.it	IP/FQDN	5061	TLS	Counte

Below the table are four configuration panels:

- Server Host:**
  - Server Lookup: IP/FQDN
  - Priority: 1
  - Host FQDN/IP: [Redacted]
  - Host IP Version: IPv4
  - Port: 5061
  - Protocol: TLS
  - TLS Profile: UC TLS Profile
- Transport:**
  - Monitor: None
- Remote Authorization and Contacts:**
  - Remote Authorization Table: None
  - Contact Registrant Table: None
  - Session URI Validation: Liberal
- Connection Reuse:**
  - Reuse: True
  - Sockets: 4
  - Reuse Timeout: Forever





The screenshot shows the Ribbon SIP settings interface. The left sidebar contains a tree view of configuration categories, with "Node-Level SIP Settings" selected and highlighted by a red box. The main panel displays the "Node-Level SIP Settings" configuration page, which includes several sections:

- Call Admission Control**: CAC Admin State is set to "Enabled".
- Source IP Untrusted Alarm**: Alarm for Untrusted INVITE is set to "Disabled".
- Skype/Lync Presence**: Presence Destination is set to "UC Front-End Servers" (highlighted by a red box), Presence Service Status is "Up" (highlighted by a green bar), and Publish Session Count is 0.
- STUN Server**: STUN Server Destination is set to "None".
- UserInfo Handling**: UserInfo Decode is set to "Legacy".

An "Apply" button is located at the bottom right of the settings panel.



## Coppia Contatto-Utente

Contatto associato al CsAnalogDevice su SfB  
Numero di telefono assegnato  
Campo Telephone number vuoto

The screenshot shows the 'Test WiFi Contact Properties' dialog box. The 'Telephones' tab is active, showing a table of attributes. The 'msRTCSIP-Line' attribute is highlighted with a red box, showing the value 'tel:+390516837807'. Below the dialog, an 'Active Directory Cache Query' window is shown with 'msRTCSIP-Line' selected as the property to match and 'tel:+390516837807' as the value to match.

Attribute	Value
cn	Test WiFi Contact
displayName	Test WiFi Contact
distinguishedName	CN=Test WiFi Contact,OU=Prese...
dSCorePropagationData	0x0 = ( )
instanceType	0x4 = ( WRITE )
msExchHideFromAddressLists	TRUE
msRTCSIP-ApplicationOptions	10
msRTCSIP-DeploymentLocator	SRV:
msRTCSIP-FederationEnabled	TRUE
msRTCSIP-InternetAccessEnabled	TRUE
msRTCSIP-Line	tel:+390516837807
msRTCSIP-OptionFlags	385
msRTCSIP-OwnerUm	um:device:analogphone
msRTCSIP-PrimaryHomeServer	CN=Lc_Services,CN=Microsoft,C...

The 'Active Directory Cache Query' window shows the following results:

Attribute	Value
Attribute=displayName	Value=Test WiFi Contact
Attribute=msRTCSIP-Line	Value=tel:+390516837807
Attribute=msRTCSIP-PrimaryUserAddress	Value=sip:testwifi.contact@centrocomputer.it

Utente abilitato su SfB  
Numero di telefono NON assegnato  
Campo Telephone number compilato

The screenshot shows the 'Test WiFi Properties' dialog box. The 'Telephones' tab is active, showing a table of attributes. The 'msRTCSIP-PrimaryUserAddress' attribute is highlighted with a red box, showing the value 'sip.test.wifi@centrocomputer.it'. Below the dialog, an 'Active Directory Cache Query' window is shown with 'telephoneNumber' selected as the property to match and '+390516837807' as the value to match.

Attribute	Value
msRTCSIP-PrimaryHomeServer	CN=Lc_Services,CN=Microsoft,C...
msRTCSIP-PrimaryUserAddress	sip.test.wifi@centrocomputer.it
msRTCSIP-UserEnabled	TRUE
msRTCSIP-UserPolicies	0=1805277387
msRTCSIP-UserRoutingGroupId	\7D\BC\0C\F9\AE\43\2D\55\8
objectCategory	CN=Person,CN=Schema,CN=Cor...
objectClass	top; person; organizationalPerson
primaryGroupId	513 = ( GROUP_RID_USERS )
proxyAddresses	sip.test.wifi@centrocomputer.it
pwdLastSet	03/04/2018 15:22:13 W. Europe
sAMAccountName	test.wifi
sAMAccountType	805306368 = ( NORMAL_USER...
telephoneNumber	+390516837807
userAccountControl	0x10200 = ( NORMAL_ACCOUN...

The 'Active Directory Cache Query' window shows the following results:

Attribute	Value
Attribute=telephoneNumber	Value=+390516837807
Attribute=msRTCSIP-PrimaryUserAddress	Value=sip:test.wifi@centrocomputer.it
Attribute=displayName	Value=Test WiFi



# Regole per Presence: Calling

Attributo	Contatto	Utente
displayName	Test WiFi Contact	Test WiFi
msRTCSIP-Line	tel:+39051***807	<null>
msRTCSIP-PrimaryUserAddress	sip:testwifi.contact@***.it	sip:test.wifi@***.it
telephoneNumber	<null>	+39051***807

ribbon Monitor Tasks Settings Diagnostics System

DA FXS A UC

Total 4 Transformation Entry Rows

Admin State	Input Field Type	Input Field Value	Output Field Type	Output Field Value	Match Type
<input type="checkbox"/>	Presence: Calling 1	==telephoneNumber== 2	Presence: Calling 4	=msRTCSIP-PrimaryUserAddress= 3	Not Applicable 5
<input type="checkbox"/>	Calling Address/Number	(.*)	Calling Address/Number	\1	Optional (Match One)
<input type="checkbox"/>	Called Address/Number	(.*)	Called Address/Number	\1	Optional (Match One)

1. Il chiamante (Calling) arriva già normalizzato in E.164 (es. +39051223344)
2. Viene fatta la query su AD per cercare un utente che abbia il chiamante nel campo telephoneNumber
3. Se la query trova un utente...
4. viene impostato lo stato "Chiamata in Corso" per l'account identificato con indirizzo sip = msRTCSIP-PrimaryUserAddress
5. Il successo della query non influisce sul successo della Transformation Rule

ribbon Monitor Tasks Settings Diagnostics System

DA REGISTRAR A UC

Total 3 Transformation Entry Rows

Admin State	Input Field Type	Input Field Value	Output Field Type	Output Field Value	Match Type
<input type="checkbox"/>	Calling Address/Number	(.*) 1	Calling Address/Number	+39051 \1 2	Optional (Match One)
<input type="checkbox"/>	Called Address/Number	(.*)	Called Address/Number	\1	Optional (Match One)
<input type="checkbox"/>	Presence: Calling 3	==telephoneNumber==	Presence: Calling 5	=msRTCSIP-PrimaryUserAddress= 4	Not Applicable 6

1. Il chiamante (Calling) arriva NON normalizzato in E.164 (es. 805 oppure 807)
2. Il chiamante viene quindi per prima cosa normalizzato
3. Viene fatta la query su AD per cercare un utente che abbia il chiamante nel campo telephoneNumber
4. Se la query trova un utente...
5. viene impostato lo stato "Chiamata in Corso" per l'account identificato con indirizzo sip = msRTCSIP-PrimaryUserAddress
6. Il successo della query non influisce sul successo della Transformation Rule



# Regole per Presence: Called

Attributo	Contatto	Utente
displayName	Test WiFi Contact	Test WiFi
msRTCSIP-Line	tel:+39051***807	<null>
msRTCSIP-PrimaryUserAddress	sip:testwifi.contact@***.it	sip:test.wifi@***.it
telephoneNumber	<null>	+39051***807

DA UC A FXS

Total 6 Transformation Entry Rows

Admin State	Input Field Type	Input Field Value	Output Field Type	Output Field Value	Match Type
<input checked="" type="checkbox"/>	Presence: Called <b>1</b>	<b>2</b> ==telephoneNumber==	Presence: Called <b>4</b>	<b>3</b> =msRTCSIP-PrimaryUserAddress=	Not Applicable <b>5</b>
<input checked="" type="checkbox"/>	Called Address/Number	\+3905 (809 907 909 911 933 934...	Called Address/Number	+3905 \1	Optional (Match One)
<input checked="" type="checkbox"/>	Called Address/Number	9003	Called Address/Number	9003	Optional (Match One)
<input checked="" type="checkbox"/>	Calling Address/Number	\+3905 ([89]\d{2})	Calling Address/Number	\1	Optional (Match One)
<input checked="" type="checkbox"/>	Calling Address/Number	\+39(.*)	Calling Address/Number	\1	Optional (Match One)
<input checked="" type="checkbox"/>	Calling Address/Number	(.*)	Calling Address/Number	\1	Optional (Match One)

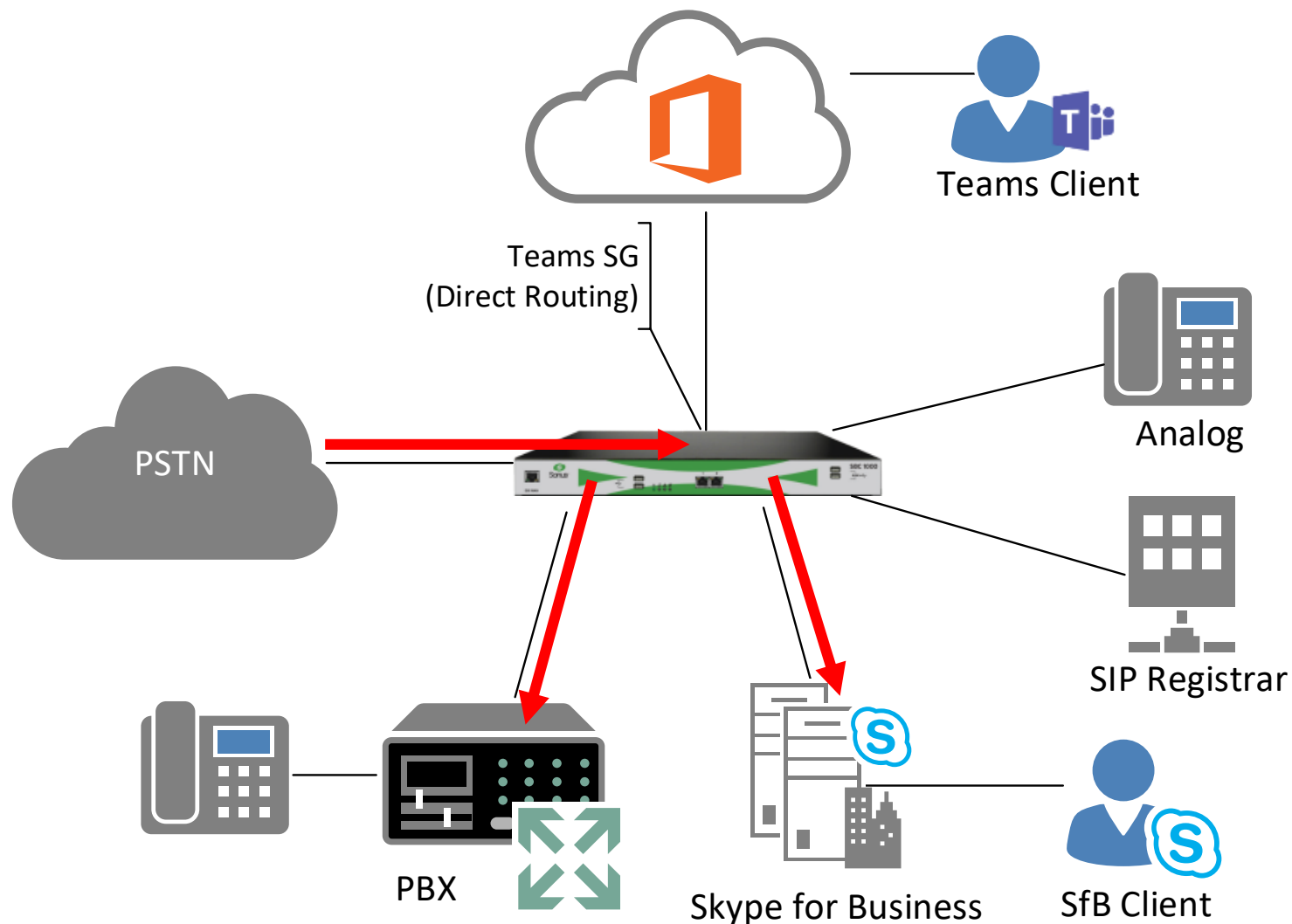
DA UC A REGISTRAR

Total 6 Transformation Entry Rows

Admin State	Input Field Type	Input Field Value	Output Field Type	Output Field Value	Match Type
<input checked="" type="checkbox"/>	Presence: Called <b>1</b>	<b>2</b> ==telephoneNumber==	Presence: Called <b>4</b>	<b>3</b> =msRTCSIP-PrimaryUserAddress=	Not Applicable <b>5</b>
<input checked="" type="checkbox"/>	Called Address/Number	\+3905 (805 807 962)	Called Address/Number	\1	Optional (Match One)
<input checked="" type="checkbox"/>	Calling Address/Number	\+3905 ([89]\d{2})	Calling Address/Number	\1	Optional (Match One)
<input checked="" type="checkbox"/>	Calling Address/Number	\+39(.*)	Calling Address/Number	\1	Optional (Match One)
<input checked="" type="checkbox"/>	Calling Address/Number	(.*)	Calling Address/Number	\1	Optional (Match One)
<input checked="" type="checkbox"/>	Calling Extension	(.*)	Calling Extension		Optional (Match One)

1. Il chiamato (Called) arriva già normalizzato in E.164 (es. +39051223344)
2. Viene fatta la query su AD per cercare un utente che abbia il # chiamato nel campo telephoneNumber
3. Se la query trova un utente...
4. viene impostato lo stato "Chiamata in Corso" per l'account identificato con indirizzo sip = msRTCSIP-PrimaryUserAddress
5. Il successo della query non influisce sul successo della Transformation Rule

## Fork della chiamata e stato di presenza



Il Forking delle chiamate è una funzione dell'SBC che consente di instradare la chiamata a più destinazioni contemporaneamente (fino a 8). Richiede una licenza di attivazione lato SBC

Risponde, positivamente, alla seguente domanda:  
Posso ricevere le chiamate sia su SfB sia sul telefono del PBX esistente, senza intervento da parte dell'utente finale?

Questa funzione si sposa perfettamente con il sistema di modifica dello stato di presenza appena visto

E' essenziale che ci sia COLLABORAZIONE da parte dei tecnici che seguono il PBX

# Fork della chiamata dalla PSTN, basato su AD e stato di presenza

Search...  
Expand All | Collapse All | Reload

**Transformation**

- DA PBX A PSTN
- DA PBX A UC
- DA PSTN A PBX
- DA PSTN A UC
- DA UC A PBX
- DA UC A PSTN
- FORK DA PBX A PBX
- FORK DA PSTN A PBX

**Call Routing Table**

- PSTN** 1
- UC
- PBX

**PSTN** October 04, 2011

Display Counters Total 4 Call Route Entry Rows

Admin State	Priority	Transformation Table	Destination Type	First Signaling Group	Description	Fork Call
2a	1	DA PSTN A UC <b>Destinazione #1</b>	Normal	(SIP) UC	Entry ID 2	Yes
2b	1	FORK DA PSTN A PBX <b>Destinazione #2</b>	Normal	(ISDN) PBX (PSTN)	Entry ID 3	No
	1	DA PSTN A UC	Deny	None	STOP CALL FORK	No
	1	DA PSTN A PBX	Normal	(ISDN) PBX (PSTN)	Entry ID 1	No

In questo esempio il Fork avviene solo se l'utente chiamato è attivo su SfB o Teams e se ha un numero a cui duplicare la chiamata in un campo riservato del suo account su AD

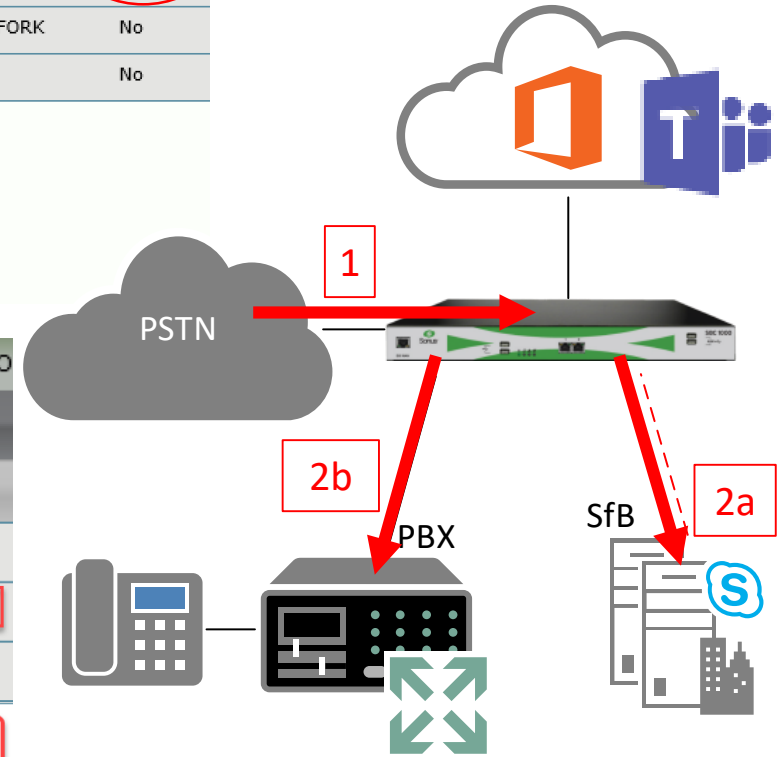
**FORK DA PSTN A PBX** 2b

Total 3 Transformation Entry Rows

Input Field Type	Input Field Value	Output Field Type	Output Field Value	Match Type
Called Address/Number	(52 \d{3}) 52 \d{3}}	User Value 1	Salvo nella variabile "User Value 1" il numero chiamato normalizzato tel:+390\1*	Optional (Match One)
User Value 1	4 Query AD =msRTCSIP-Line=	6 Called Address/Number	=extensionAttribute9=	5 Mandatory (Must Match)
Presence: Called	==extensionAttribute9==	7 Presence: Called	=msRTCSIP-PrimaryUserAddress=	No, Applicable

Imposto lo stato di presenza dell'account AD chiamato su "Chiamata in corso"

Perchè il fork avvenga deve essere presente un numero da chiamare sul PBX in questo campo riservato su AD



Fork della chiamata dal PBX, basato su AD e stato di presenza per chiamato e chiamante

Search...

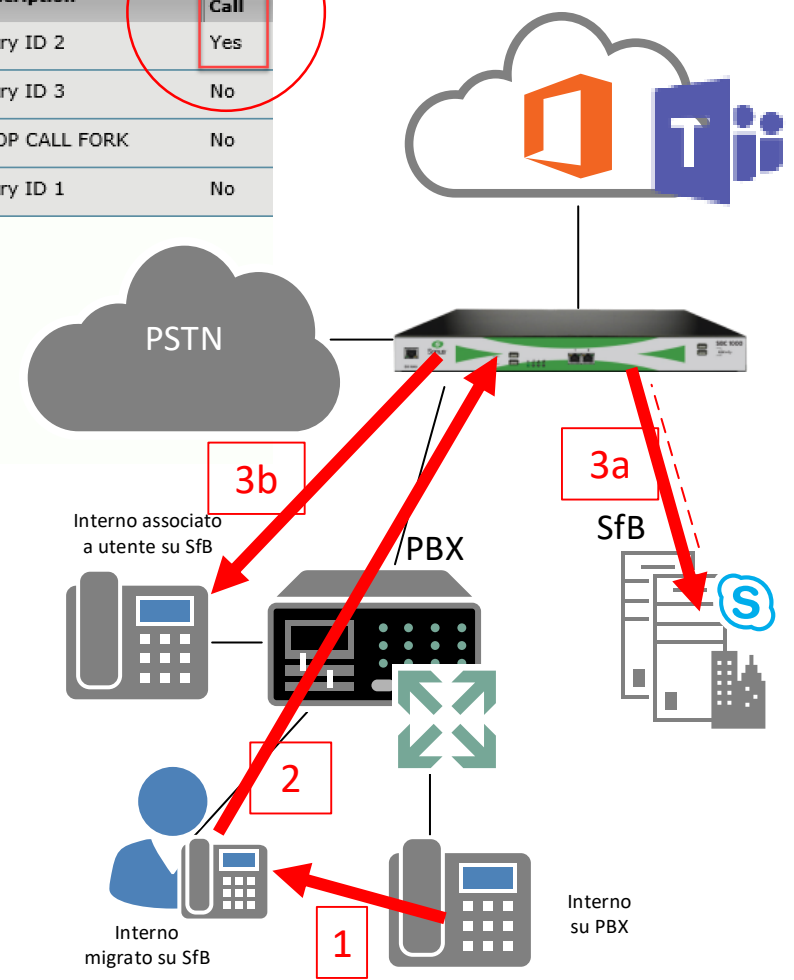
Expand All | Collapse All | Reload

- Transformation
  - DA PBX A PSTN
  - DA PBX A UC
  - DA PSTN A PBX
  - DA PSTN A UC
  - DA UC A PBX
  - DA UC A PSTN
  - FORK DA PBX A PBX
  - FORK DA PSTN A PBX
- Call Routing Table
  - PSTN
  - UC
  - PBX 2**
- Call Actions

**PBX** October 04, 2012

Display Counters Total 4 Call Route Entry Rows

Admin State	Priority	Transformation Table	Destination Type	First Signaling Group	Description	Fork Call
<input type="checkbox"/> <input checked="" type="checkbox"/>	1 <b>3a</b>	DA PBX A UC <b>Destinazione 1</b>	Normal	(SIP) UC	Entry ID 2	Yes
<input type="checkbox"/> <input checked="" type="checkbox"/>	1 <b>3b</b>	FORK DA PBX A PBX <b>Destinazione 2</b>	Normal	(ISDN) PBX (UC)	Entry ID 3	No
<input type="checkbox"/> <input checked="" type="checkbox"/>	1	DA PBX A UC	Deny	None	STOP CALL FORK	No
<input type="checkbox"/> <input checked="" type="checkbox"/>	1	DA PBX A PSTN	Normal	(ISDN) PSTN	Entry ID 1	No



# Fork della chiamata dal PBX, basato su AD e stato di presenza per chiamato e chiamante

**FORK DA PBX A PBX** **3b**

Total 14 Transformation

Input Field Type	Input Field Value	Output Field Type	Output Field Value	Match Type
Called Address/Number	(^[35]\d{3}\$)	User Value 1	tel:\1*	Optional (Match One)
Called Address/Number	(05\d{3} 05\d{3})	User Value 1	tel:+39\1*	Optional (Match One)
User Value 1	=msRTCSIP-Line=	Called Address/Number	=extensionAttribute9=	Mandatory (Must Match)
Presence: Called	=extensionAttribute9=	Presence: Called	=msRTCSIP-PrimaryUserAddress=	Not Applicable
Calling Address/Number	(^[35]\d{3}\$)	User Value 2	\1	Optional (Match One)
Calling Address/Number	(52\d{3} 52\d{3})	User Value 2	+390\1	Optional (Match One)
User Value 2	(.*)	User Value 2	\1	Optional (Match One)
User Value 2	==telephoneNumber==	Presence: Calling	=msRTCSIP-PrimaryUserAddress=	Optional (Match One)
Calling Address/Number	52\d{3}	Calling Address/Number	2\1	Optional (Match One)
Calling Address/Number	52\d{3}	Calling Address/Number	4\1	Optional (Match One)
Calling Address/Number	(.*)	Calling Name	\1	Optional (Match One)
User Value 2	==telephoneNumber==	Calling Name	=displayName=	Optional (Match One)
Calling Address/Number	(.*)	Calling Numbering Plan	Private	Optional (Match One)
Calling Address/Number	(.*)	Calling Numbering Type	Unknown	Optional (Match One)

Nella variabile User Value 1 viene salvato il numero **chiamato** normalizzato. L'operazione viene ripetuta due volte perchè in questo esempio il cliente ha due numerazioni (passante e non passante)

Perchè il Fork avvenga deve essere presente un numero da chiamare sul PBX in questo campo AD riservato sull'account utente chiamato

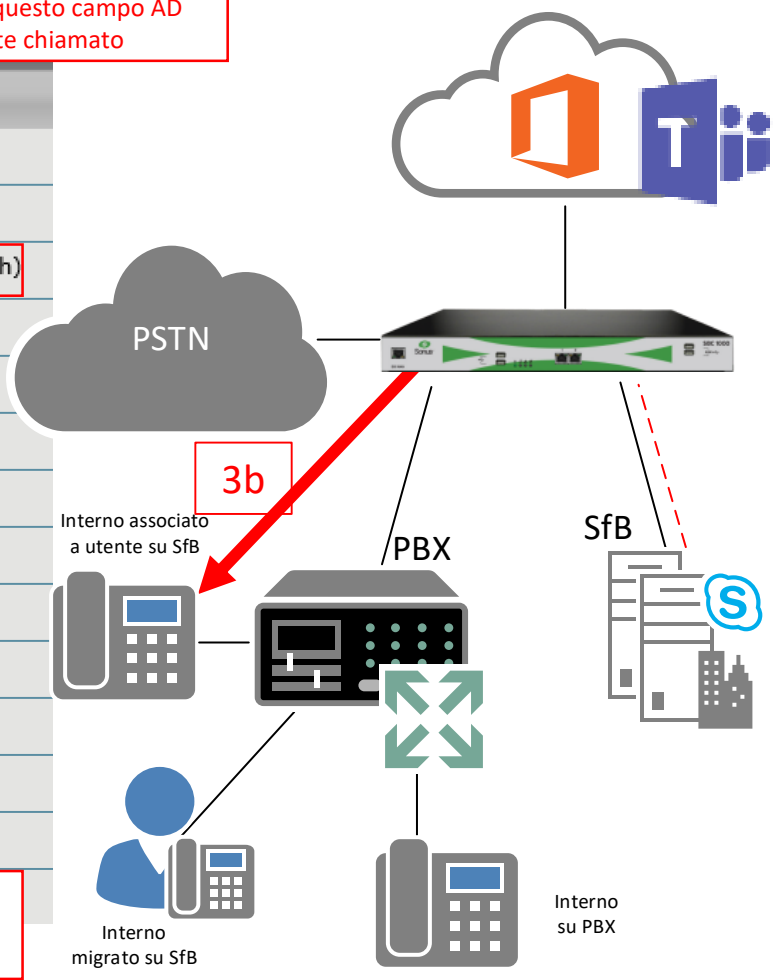
Nella variabile User Value 2 viene salvato il numero **chiamante** normalizzato

Safe Rule: fondamentale!

Presento il #chiamante con il suo interno

Se disponibile in AD viene inserito il Display Name del chiamante nel campo SIP Calling Name

Se il PBX lo supporta, vengono modificati i valori NPI e TON per presentare la chiamata come interna al PBX stesso





## Conclusioni

Abbiamo visto esempi di servizi avanzati legati alla fonia, basati sul componente chiave Voice Gateway

- SIP Registrar
- Status Presence
- Call Fork

Questi progetti richiedono:

- Idee chiare su cosa si vuole (e cosa si può) ottenere
- Esperienza e competenza su tutti i prodotti interessati
- Collaborazione da parte dei gestori dei PBX esistenti

