

Mengintegrasikan VoIP dengan 3CX Menggunakan CUCM

Alkindi Hafidz

Alkindi.h@outlook.com

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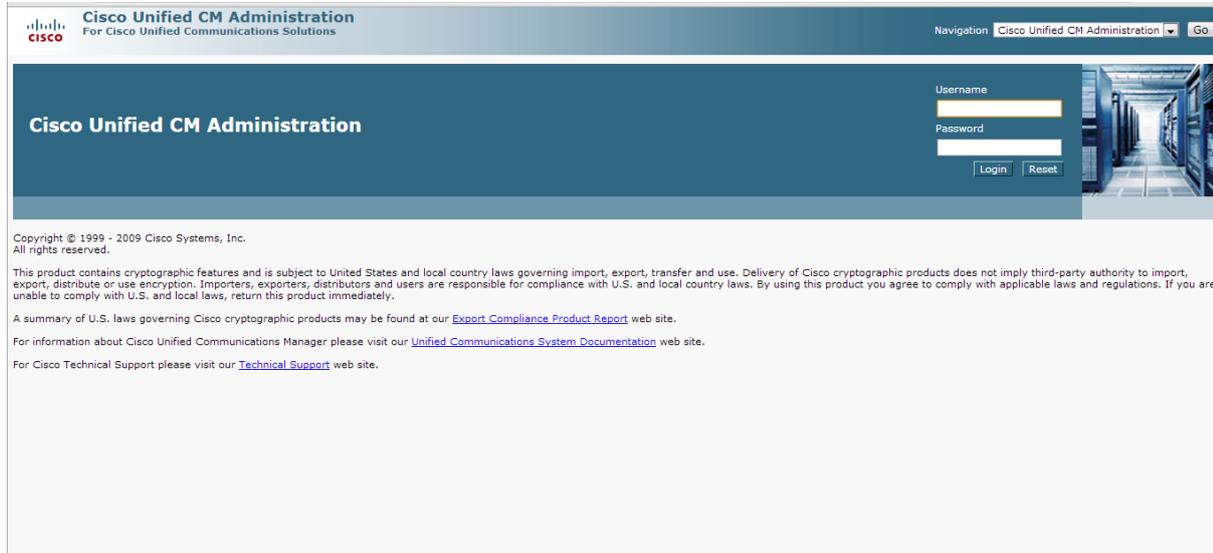
Cisco Unified Communications Manager (CUCM), sebelumnya Cisco Unified CallManager dan Cisco CallManager (CCM) adalah sistem pengolah panggilan berbasis software yang dikembangkan oleh Cisco Systems. CUCM melacak semua komponen jaringan VoIP yang aktif, termasuk ponsel, gateway, conference bridge, sumber daya transcoding, dan kotak pesan suara diantara yang lainnya. Call Manager sering memanfaatkan Skinny Client Control Protocol (SCCP) sebagai protokol komunikasi untuk menandakan titik akhir hardware dari sistem, seperti IP Phones. H.323, Media Gateway Control Protocol (MGCP) atau Session Initiation Protocol (SIP) digunakan untuk melewati sinyal panggilan ke gateway. Cisco Unified Communications Manager menyediakan beberapa fungsionalitas tertentu untuk klien Cisco Mobile VoIP yang terhubung langsung dengan Cisco Unified Communications Manager.

3CX Phone System adalah sebuah software IP PBX yang dapat menggantikan perangkat fisik PBX / PABX. IP PBX dari 3CX ini telah didevelop khusus untuk sistem operasi Windows dan berprotokol standar SIP, sehingga akan lebih mudah dimanage dan tentunya akan cocok dengan segala jenis SIP Phone, softphone maupun IP Phone. 3CX Phone System, selain berbasis Windows, 3CX juga memberikan paket teknologi komunikasi yang lengkap dengan menyertakan voice mail, fax, email dan status kehadiran/online user dan juga Video Call.

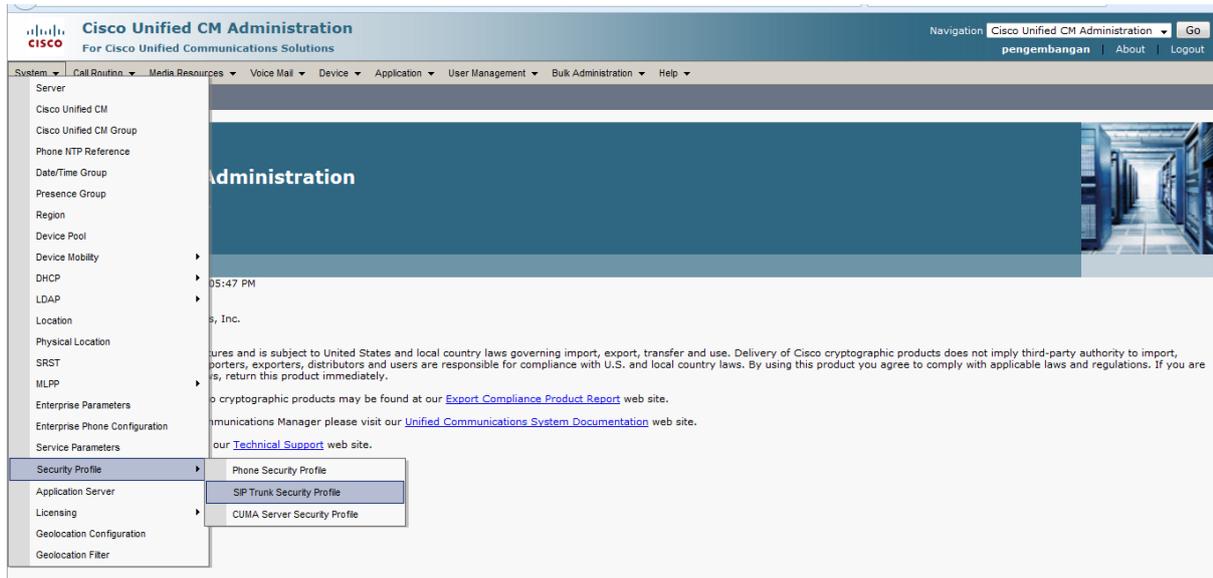
CUCM dapat diintegrasikan dengan VoIP IPPhone yang lain dengan menggunakan protocol SIP, dalam artikel ini, penulis akan mencoba mengintegrasikan CUCM dengan VoIP IPPhone 3CX. Berikut adalah langkah – langkah lengkap untuk mengintegrasikan 3CX dengan CUCM.

Langkah – langkahnya mengintegrasikan 3CX dengan CUCM :

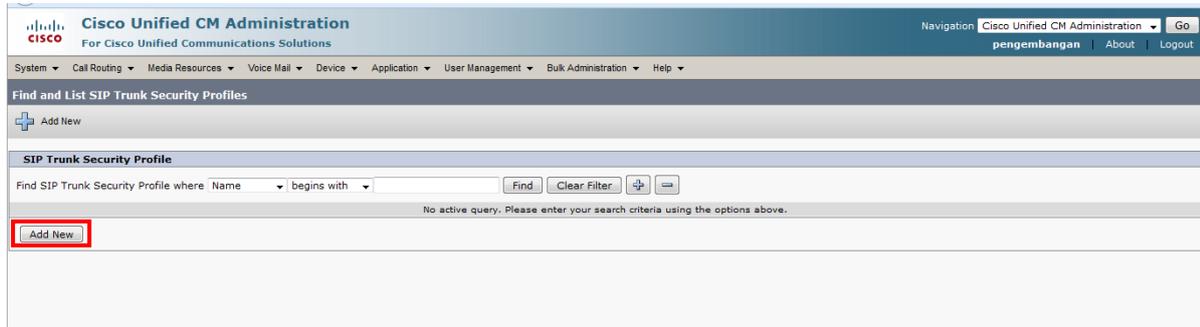
1. Langkah pertama adalah membuat konfigurasi SIP Trunk antara 3CX dengan CUCM
 - a. Masuk ke Server CUCM anda :



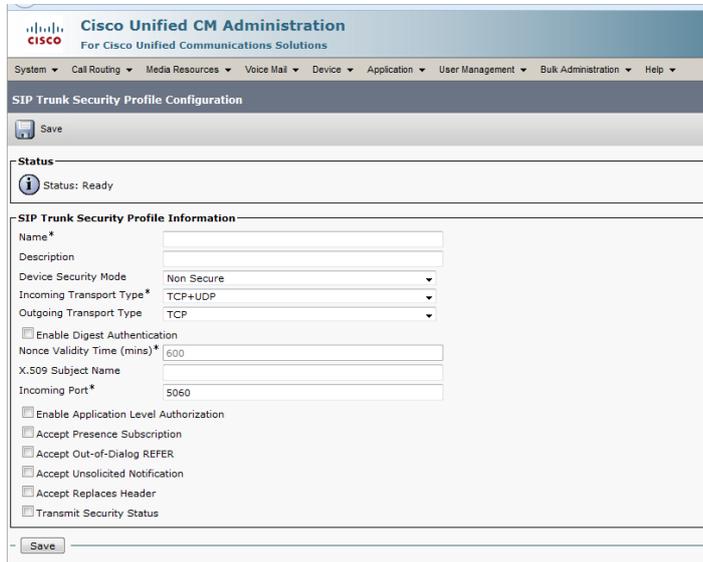
- b. Lalu masuk ke **Menu System – Security**, lalu pilih **SIP Trunk Security Profile** :



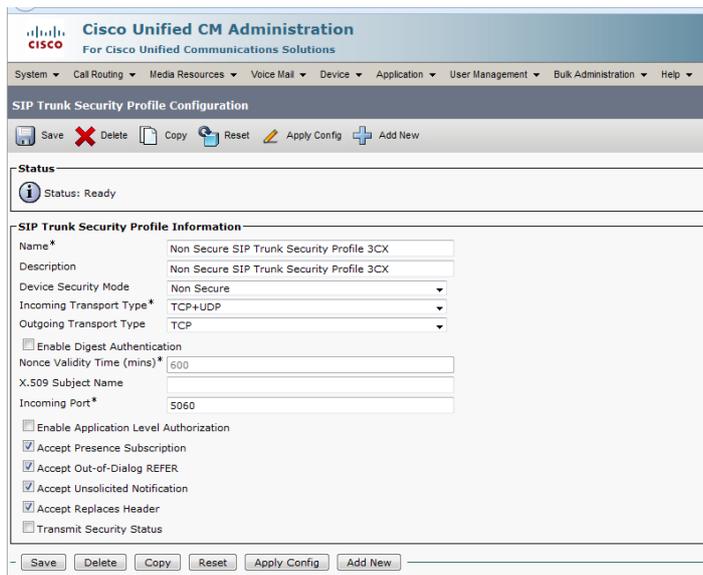
c. Setelah tampil menu seperti ini, lalu pilih **Add New** :



d. Isi pada form Add Menu ini, dengan data-data yang diperlukan seperti pada gambar dibawahnya :



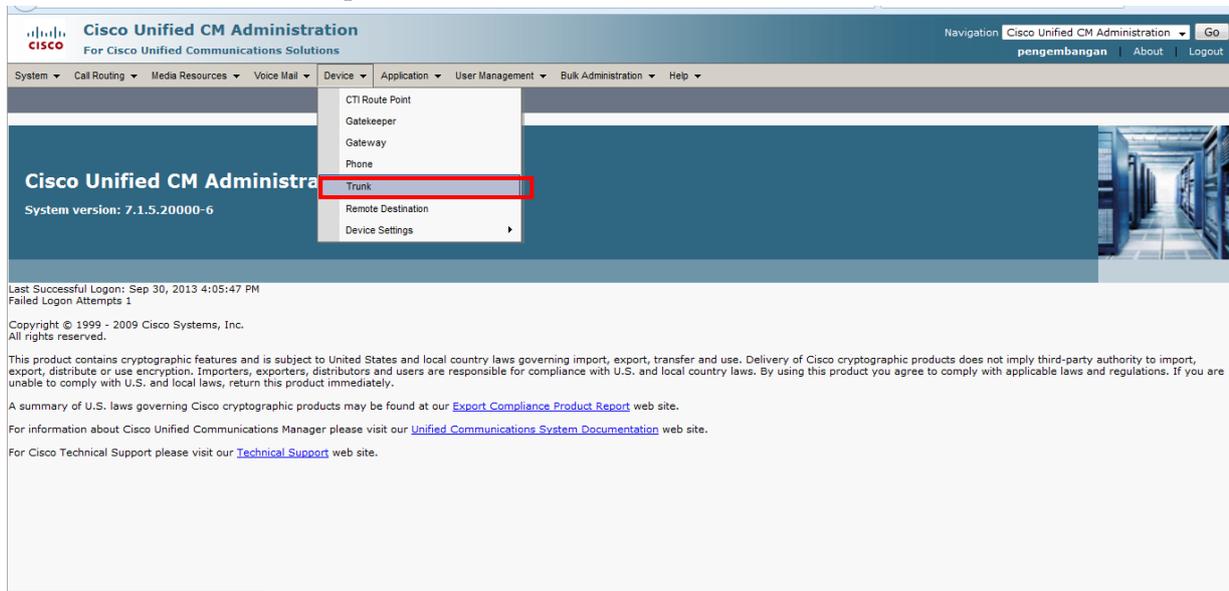
Contoh :



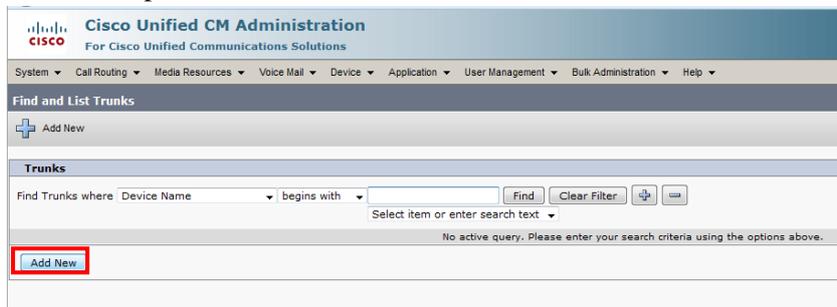
Setelah mengisi semua datanya, lakukan save.

Selanjutnya kita buat konfigurasi Trunk untuk mengarahkan Trunk ke IP server 3CX.

e. Masuk ke menu **Device** lalu pilih **Trunk** :



Setelah tampilan ini terbuka, lakukan **Add New** :



Pilih pada **Trunk Type** nya : “SIP Trunk” dan Device Protokolnya akan secara otomatis mengikuti pilihan kita pada saat memilih Trunk Type. Setelah itu pilih Next :

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Trunk Configuration

Next

Status
Status: Ready

Trunk Information

Trunk Type* SIP Trunk
Device Protocol* SIP

Next

*. indicates required item.

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Trunk Configuration Related Links: Back To Find/List ▾

Save

Status
Status: Ready

Device Information

Product:	SIP Trunk
Device Protocol:	SIP
Device Name*	
Description	
Device Pool*	-- Not Selected --
Common Device Configuration	< None >
Call Classification*	Use System Default
Media Resource Group List	< None >
Location*	Hub_None
AAR Group	< None >
Packet Capture Mode*	None
Packet Capture Duration	0
<input type="checkbox"/> Media Termination Point Required	
<input checked="" type="checkbox"/> Retry Video Call as Audio	
<input type="checkbox"/> Transmit UTF-8 for Calling Party Name	
<input type="checkbox"/> Unattended Port	
<input type="checkbox"/> SRTP Allowed - When this flag is checked, Encrypted TLS needs to be configured in the network to provide end to end security. Failure to do so will expose keys and other information.	
Use Trusted Relay Point*	Default

Lalu isi sesuai contoh dibawah ini :

Buat Trunk 3CX, dengan mengarahkan trunk ke IP Trunk Server 3CX :

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Trunk Configuration Related Links: Back To Find/List **Go**

Save Delete Reset Apply Config Add New

Status
 Status: Ready

Device Information

Product: SIP Trunk
 Device Protocol: SIP
 Device Name*:
 Description: TRUNK_3CX
 Device Pool*: DP_JKT
 Common Device Configuration: < None >
 Call Classification*: OnNet
 Media Resource Group List: < None >
 Location*: Jakarta
 AAR Group: < None >
 Packet Capture Mode*: Batch Processing Mode
 Packet Capture Duration: 0

Media Termination Point Required
 Retry Video Call as Audio
 Transmit UTF-8 for Calling Party Name
 Unattended Port
 SRTP Allowed - When this flag is checked, Encrypted TLS needs to be configured in the network to provide end to end security. Failure to do so will expose keys and other information.
 Use Trusted Relay Point*: Default

Incoming Calling Party Settings

If the administrator sets the prefix to Default this indicates call processing will use prefix at the next level setting (DevicePool/Service Parameter). Otherwise, the value configured is used as the prefix unless the field is empty in which case there is no prefix assigned.

Clear Prefix Settings Default Prefix Settings

Number Type	Prefix	Strip Digits	Use Device Pool CSS	Calling Search Space
Unknown Number	Default	0	<input checked="" type="checkbox"/>	< None >

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Trunk Configuration Related Links: Back To Find/List **Go**

Save Delete Reset Apply Config Add New

Incoming Calling Party Settings

If the administrator sets the prefix to Default this indicates call processing will use prefix at the next level setting (DevicePool/Service Parameter). Otherwise, the value configured is used as the prefix unless the field is empty in which case there is no prefix assigned.

Clear Prefix Settings Default Prefix Settings

Number Type	Prefix	Strip Digits	Use Device Pool CSS	Calling Search Space
Unknown Number	Default	0	<input checked="" type="checkbox"/>	< None >

Multilevel Precedence and Preemption (MLPP) Information
 MLPP Domain: < None >

Call Routing Information

Remote-Party-Id
 Asserted-Identity
 Asserted-Type*: Default
 SIP Privacy*: Default

Inbound Calls

Significant Digits*: All
 Connected Line ID Presentation*: Default
 Connected Name Presentation*: Default
 Calling Search Space: CSS_Global
 AAR Calling Search Space: CSS_Global
 Prefix DN:
 Redirecting Diversion Header Delivery - Inbound

Outbound Calls

Called Party Transformation CSS: CSS_Global
 Use Device Pool Called Party Transformation CSS
 Calling Party Transformation CSS: CSS_Global

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Trunk Configuration Related Links: Back To Find/List

Save Delete Reset Apply Config Add New

Outbound Calls

Called Party Transformation CSS: CSS_Global
 Use Device Pool Called Party Transformation CSS
Calling Party Transformation CSS: CSS_Global
 Use Device Pool Calling Party Transformation CSS
Calling Party Selection*: Originator
Calling Line ID Presentation*: Default
Calling Name Presentation*: Default
Caller ID DN:
Caller Name:
 Redirecting Diversion Header Delivery - Outbound

SIP Information

Destination Address: 10.100.93.172
Destination Address IPv6:
 Destination Address is an SRV
Destination Port*: 5060
MTP Preferred Originating Codec*: 711ulaw
Presence Group*: Standard Presence group
SIP Trunk Security Profile*: Non Secure SIP Trunk Security Profile 3CX
Rerouting Calling Search Space: CSS_Global
Out-Of-Dialog Refer Calling Search Space: CSS_Global
SUBSCRIBE Calling Search Space: CSS_Global
SIP Profile*: Standard SIP Profile
DTMF Signaling Method*: No Preference

Geolocation Configuration

Geolocation: < None >
Geolocation Filter: < None >

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Save Delete Reset Apply Config Add New

Calling Line ID Presentation: Default
Calling Name Presentation*: Default
Caller ID DN:
Caller Name:
 Redirecting Diversion Header Delivery - Outbound

SIP Information

Destination Address: 1
Destination Address IPv6:
 Destination Address is an SRV
Destination Port*: 5060
MTP Preferred Originating Codec*: 711ulaw
Presence Group*: Standard Presence group
SIP Trunk Security Profile*: Non Secure SIP Trunk Security Profile 3CX
Rerouting Calling Search Space: CSS_Global
Out-Of-Dialog Refer Calling Search Space: CSS_Global
SUBSCRIBE Calling Search Space: CSS_Global
SIP Profile*: Standard SIP Profile
DTMF Signaling Method*: No Preference

Geolocation Configuration

Geolocation: < None >
Geolocation Filter: < None >
 Send Geolocation Information

Save Delete Reset Apply Config Add New

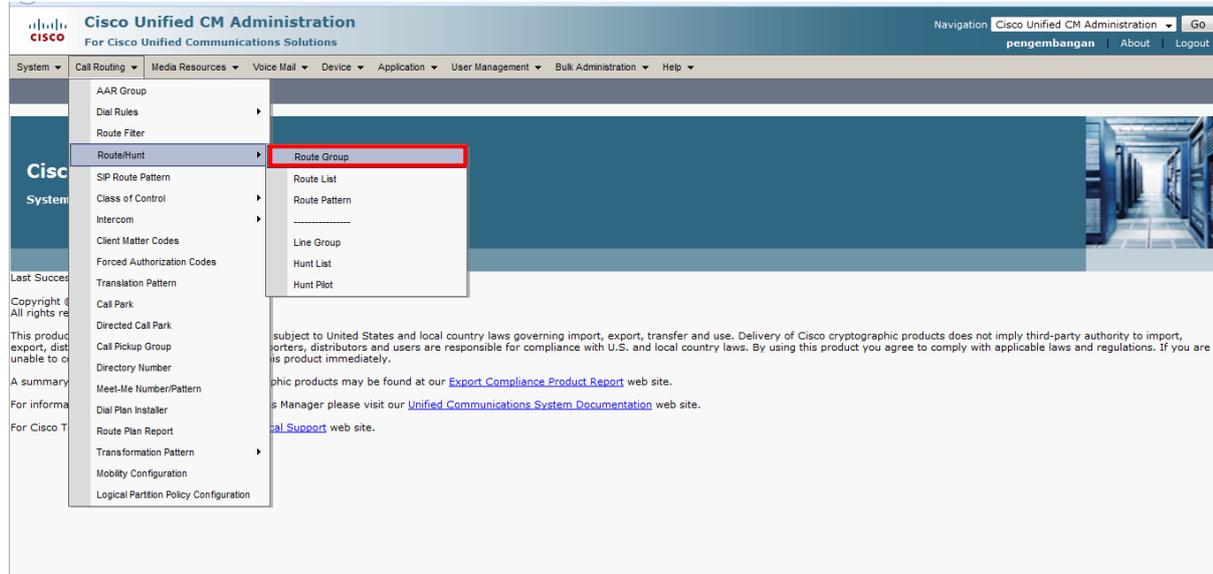
Destination Address ini arahkan ke IP Address Server 3CX

Pilih SIP Trunk Security Profile yang telah dibuat sebelumnya

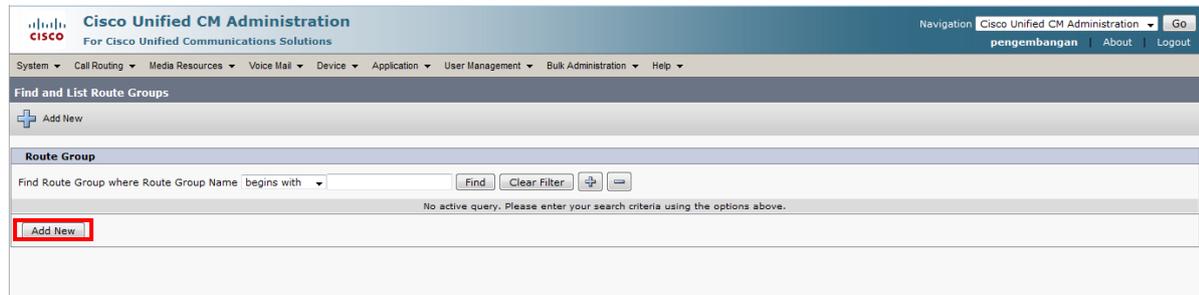
*- indicates required item.
**- Device reset is not required for changes to Packet Capture Mode and Packet Capture Duration.

Setelah data-data selesai diisi, jangan lupa lakukan save.

2. Setelah selesai membuat konfigurasi Trunk, langkah selanjutnya adalah membuat konfigurasi pengaturan digit. Ada beberapa langkah dalam konfigurasi pengaturan digit ini, yaitu :
 - a. **Route Group** yang berfungsi Route Group kita buat untuk mengarahkan pattern yang nantinya kita buat. Langkah-langkahnya adalah sebagai berikut :
 - ❖ Masuk ke Menu **Call Routing – Route/Hunt**, lalu pilih **Route Group** :



- ❖ Setelah tampilan ini terbuka, pilih Add New :



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Route Group Configuration

Save

Route Group Information

Route Group Name*

Distribution Algorithm* Circular ▾

Route Group Member Information

Find Devices to Add to Route Group

Device Name contains Find

Available Devices**

Port(s) All ▾

Add to Route Group

Current Route Group Members

Selected Devices***

Reverse Order of Selected Devices

Removed Devices****

- ❖ Setelah tampilan diatas terbuka, isi data-datanya sesuai dengan template dibawah ini Seharusnya, jika konfigurasi Trunk kita tadi diawal berhasil, maka pada Available device akan muncul alamat IP Address server 3CX.

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Route Group Configuration Related Links: Back To Find/List ▾ Go

Save Delete Add New

Route Group Information

Route Group Name* RG_3CX

Distribution Algorithm* Circular ▾

Route Group Member Information

Find Devices to Add to Route Group

Device Name contains Find

Available Devices**

Port(s) All ▾

Add to Route Group

Current Route Group Members

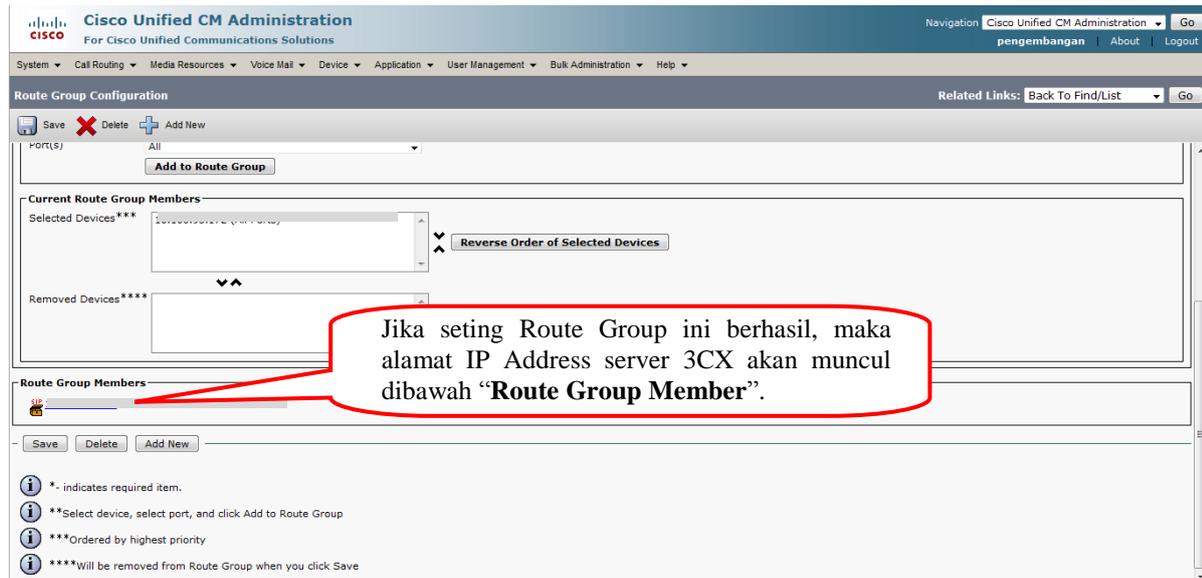
Selected Devices***

Reverse Order of Selected Devices

Removed Devices****

Jangan lupa Klik Add to Route Group setelah memilih IP Address server 3CX anda pada Available Device.

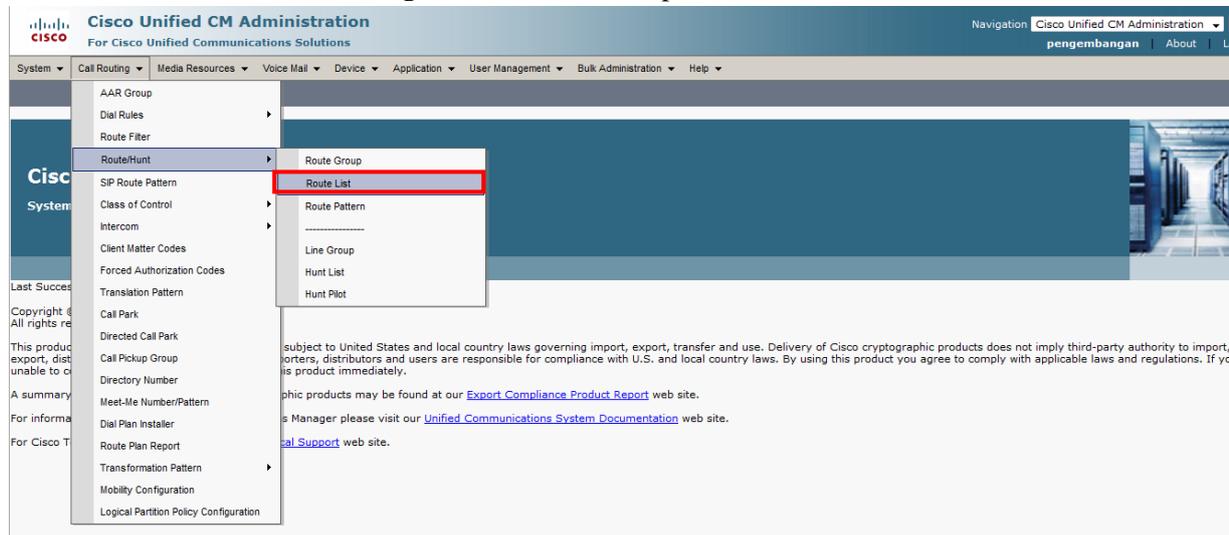
Setelah Add to Route Group diklik, maka IP address server 3CX akan muncul disini



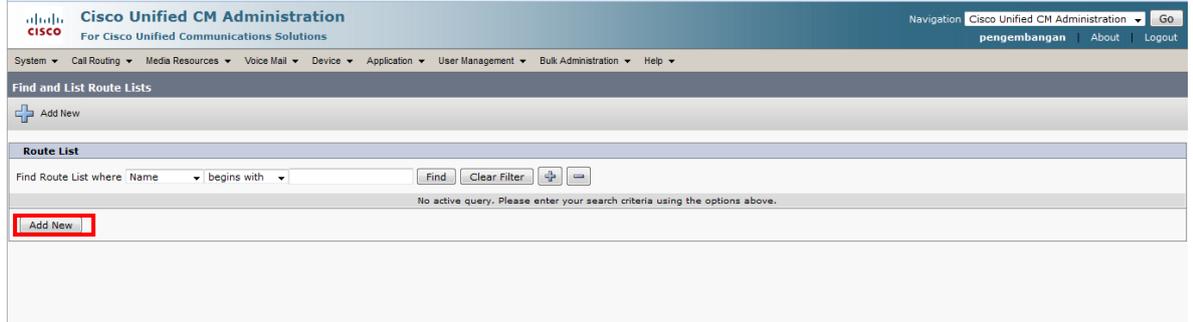
❖ Setelah selesai mengkonfigurasi Route Group, maka lakukan Save.

b. Setelah Route Group selesai dikonfigurasi, selanjutnya kita harus membuat **Route list** yang berfungsi sebagai jembatan dari Route Pattern ke Route Group. Langkah – langkahnya adalah sebagai berikut :

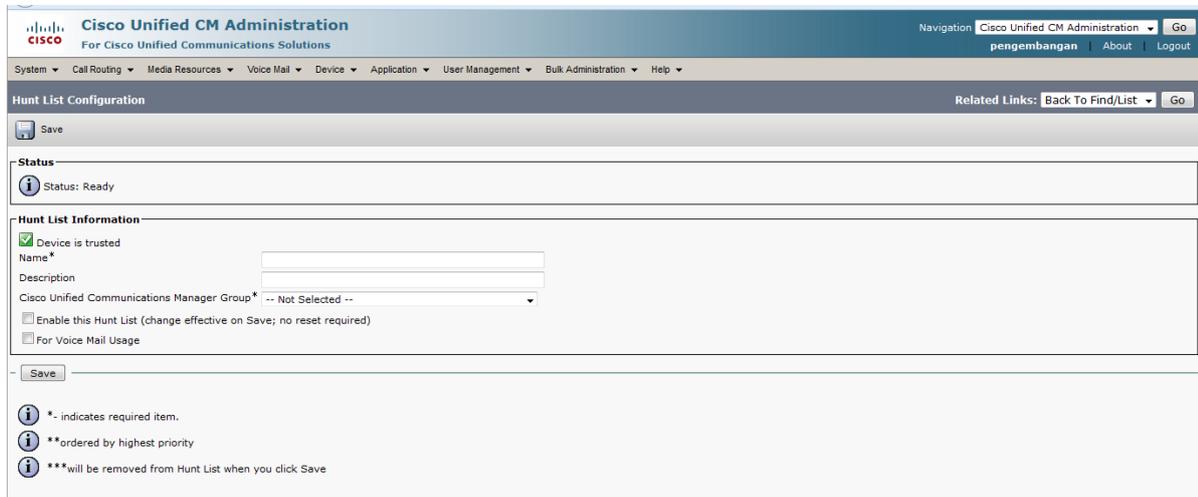
❖ Masuk ke Menu **Call Routing – Route/Hunt**, lalu pilih **Route List**



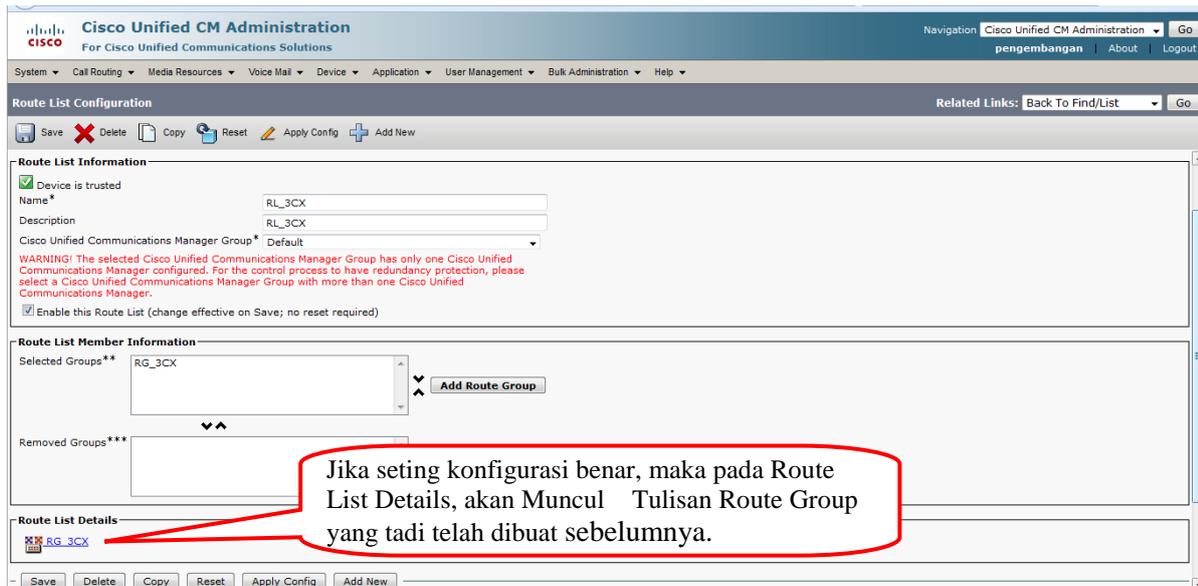
❖ Setelah tampilan dibawah terbuka, pilih **Add New** :



❖ Setelah tampilan ini terbuka, isi data-data yang dibutuhkan seperti contoh dibawah :



Contoh :

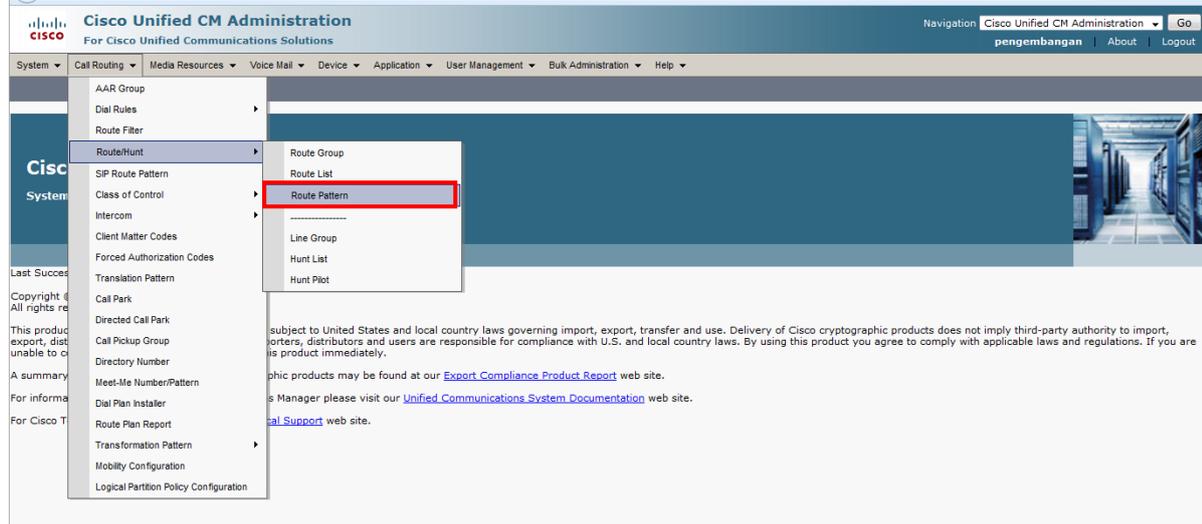


Jika seting konfigurasi benar, maka pada Route List Details, akan Muncul Tulisan Route Group yang tadi telah dibuat sebelumnya.

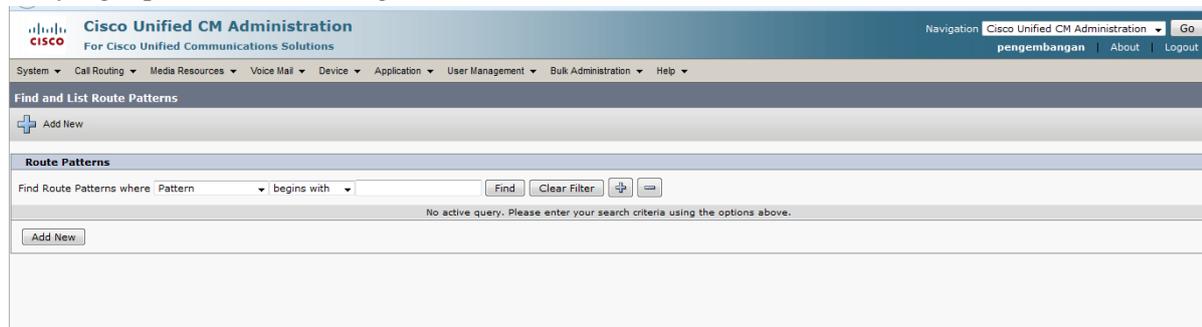
Setelah selesai mengisi data-data yang diperlukan, lakukan save.

- c. Setelah selesai membuat Route List, langkah selanjutnya adalah membuat **Route Pattern** yang berfungsi untuk memetakan digit yang diterima oleh CUCM. Langkah-langkahnya adalah sebagai berikut :

❖ Masuk ke Menu **Call Routing – Route/Hunt**, lalu pilih **Route Pattern** :



❖ Setelah tampilan dibawah ini terbuka, Pilih **Add New** kemudian isi data-data yang diperlukan sesuai dengan contoh dibawah ini :



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Route Pattern Configuration | Related Links: Back To Find/List | Go

Save

Status: Ready

Pattern Definition

Route Pattern*
Route Partition: < None >
Description
Numbering Plan: -- Not Selected --
Route Filter: < None >
MLPP Precedence*: Default
Resource Priority Namespace Network Domain: < None >
Gateway/Route List*: -- Not Selected -- (Edit)
Route Option: Route this pattern, Block this pattern No Error

Call Classification*: OffNet

Allow Device Override Provide Outside Dial Tone Allow Overlap Sending Urgent Priority
 Require Forced Authorization Code
Authorization Level*: 0
 Require Client Matter Code

Calling Party Transformations

Use Calling Party's External Phone Number Mask
Calling Party Transform Mask

❖ Contoh :

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Route Pattern Configuration | Related Links: Back To Find/List | Go

Save X Delete Copy Add New

Status: Ready

Pattern Definition

Route Pattern*: 3XXX
Route Partition: P
Description: To 3CX
Numbering Plan: -- Not Selected --
Route Filter: < None >
MLPP Precedence*: Default
Resource Priority Namespace Network Domain: < None >
Gateway/Route List*: RL_3CX (Edit)
Route Option: Route this pattern, Block this pattern No Error

Call Classification*: OnNet

Allow Device Override Provide Outside Dial Tone Allow Overlap Sending Urgent Priority
 Require Forced Authorization Code
Authorization Level*: 0
 Require Client Matter Code

Calling Party Transformations

Use Calling Party's External Phone Number Mask
Calling Party Transform Mask

Route Pattern 3XXX ke Arah 3CX.

Pada Route Patternnya, isi dengan angka yang diinginkan, dalam artikel ini, penulis memilih angka 3XXX untuk route pattern ke 3CX. Setelah selesai mengisi data-data yang diperlukan, lakukan save. Dengan melakukan save, maka selesailah kita dalam mengintegrasikan 3CX dengan CUCM.

Demikianlah tutorial singkat Mengintegrasikan VoIP dengan 3CX dengan CUCM. Semua device yang menggunakan protocol SIP yang akan diintegrasikan dengan CUCM dapat merujuk ke artikel ini karena langkah-langkah pengintegrasian hampir sama. Mohon maaf apabila dalam penulisan artikel terdapat ketidakjelasan dan menimbulkan pertanyaan, semua karena keterbatasan penulis, insyaAllah akan berusaha belajar lagi. Semoga dapat bermanfaat bagi penulis sendiri dan teman-teman semua. ☺

Referensi :

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Biografi Penulis



Alkindi Hafidz.

Menyelesaikan S1 di Universitas Muhammadiyah Malang Jurusan Teknik Informatika lulus tahun 2009. Berminat dengan dunia networking terutama Voip.