

## UCx to IPO Sip Trunk

UC<sup>x</sup> IP Address for this example is 192.168.1.200

IPO IP Address for this example is 192.168.1.250

Login to IPO

1. Right Click on Line
2. Select New
3. Choose SIP Line Tab
4. Under the SIP line Tab
  - a. Choose Line number that you want to name this line. (Ex: 18)
5. Fill in the Tabs on the right SIP entry with the following screen shots.

The screenshot shows a configuration window titled "SIP Line - Line 18". The window has several tabs: "SIP Line", "Transport", "SIP URI", "VoIP", "T3E Fax", and "SIP Credentials". The "SIP Line" tab is active. The configuration is as follows:

Field	Value	Field	Value
Line Number	18	In Service	<input checked="" type="checkbox"/>
ITSP Domain Name	192.168.1.200	Use Tel URI	<input type="checkbox"/>
Prefix		Check OOS	<input checked="" type="checkbox"/>
National Prefix	0	Call Routing Method	Request URI
Country Code		Originator number for forwarded and twinning calls	
International Prefix	00	Name Priority	System Default
Send Caller ID	Diversion Header	Caller ID from From header	<input type="checkbox"/>
Association Method	By Source IP address	Send From In Clear	<input type="checkbox"/>
<input checked="" type="checkbox"/> REFER Support		User-Agent and Server Headers	
Incoming	Auto		
Outgoing	Auto		
UPDATE Supported	Never		

### SIP Line - Line 18

SIP Line Transport SIP URI VoIP T38 Fax SIP Credentials

ITSP Proxy Address

**Network Configuration**

Layer 4 Protocol  Send Port

Use Network Topology Info  Listen Port

Explicit DNS Server(s)

Calls Route via Registrar

Separate Registrar

### SIP Line - Line 18

SIP Line Transport SIP URI VoIP T38 Fax SIP Credentials

Channel	Groups	Via	Local URI	Contact	Display Name	PAI	Credential	Max Calls
1	18 18	<...					0: <Non...	10

### SIP Line - Line 18

SIP Line Transport SIP URI VolP T38 Fax SIP Credentials

Codec Selection: System Default

Unused

>>>

↑

<<<

↓

>>>

Selected

- G.711 ULAW 64K
- G.711 ALAW 64K
- G.729(a) 8K CS-ACELP
- G.723.1 6K3 MP-MLQ

Fax Transport Support: None

Call Initiation Timeout (s): 4

DTMF Support: RFC2833

VoIP Silence Suppression

Re-invite Supported

Use Offerer's Preferred Codec

Codec Lockdown

PRACK/100rel Supported

### SIP Line - Line 18

SIP Line Transport SIP URI VolP T38 Fax SIP Credentials

T38 Fax Version: 3

Transport: UDPTL

Redundancy:

Low Speed: 0

High Speed: 0

TCF Method: Trans TCF

Max Bit Rate (bps): 14400

EFlag Start Timer (msecs): 2600

EFlag Stop Timer (msecs): 2300

Tx Network Timeout (secs): 150

Scan Line Fix-up

TFOIP Enhancement

Disable T30 ECM

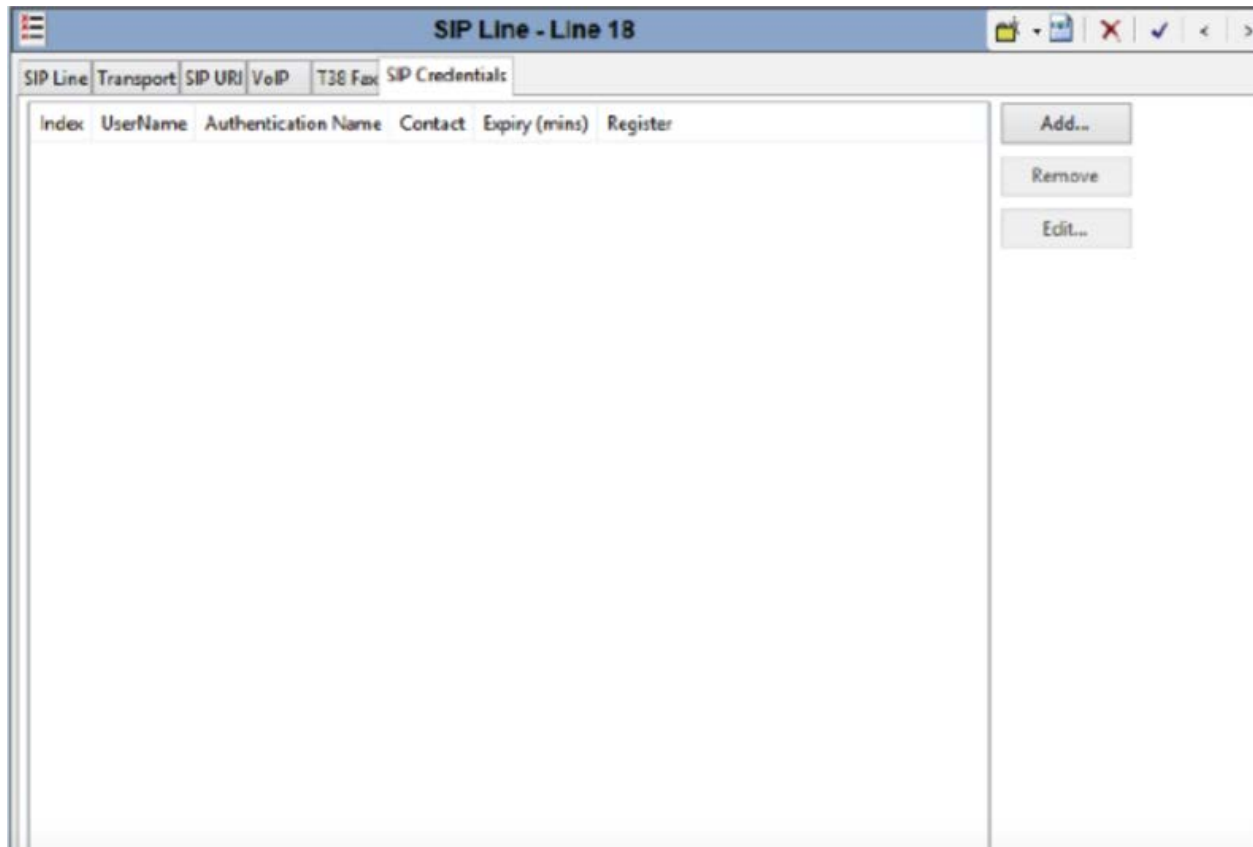
Disable EFlags For First DIS

Disable T30 MR Compression

NSF Override

Country Code: 0

Vendor Code: 0



Login to the UC<sup>x</sup>

1. Go to PBX tab
2. Click Trunks on left menu
3. Enter Trunk Name, then submit and apply

**E-MetroTel**

System Fax **PBX** Reports Extras My Extension Security Support

PBX Configuration Operator Panel Voicemails Call Recordings Batch Configuration Conference Tools Trunk Providers MDSE

**PBX Configuration**

Basic

- Extensions
- Extensions (Nortel)
- Feature Codes
- Numbering Plan
- Trunks**

Outbound Call Control

- Outbound Routes
- Congestion Messages
- Custom Contexts

Inbound Call Control

- Inbound Routes
- Announcements
- Blacklist
- CallerID Lookup Sources
- Call Flow Control
- Call Recording
- Follow Me
- IVR
- PSTN Trunk DIDs
- Queue Priorities
- Queues
- Ring Groups
- Set CallerID
- Time Conditions
- Time Groups

Settings

- Advanced Settings
- Fax Settings
- IAX Settings
- Log File Settings
- Nortel Settings
- SIP Settings
- Survivability Settings
- Voicemail Settings

**Edit SIP Trunk**

Delete Trunk IPOffice

In use by 1 route

General Settings

Trunk Name: IPOffice

Outbound CallerID:

CID Options: Allow Any CID

Maximum Channels:

Asterisk Trunk Dial Options: tTwW  Override

Continue if Busy:  Check to always by next trunk

Disable Trunk:  Disable

Dial Number Manipulation Rules

(prepend) + prefix | match pattern

+ Add More Dial Pattern Fields Clear all Fields

Dial Rules Wizards: (pick one)

Outbound Dial Prefix:

Add Trunk

- emetrotel (fax)
- Vincent (fax)
- BCM (sip)
- Colleen Edgren (sip)
- Dan Gerneau (sip)
- IPOffice (sip)
- Jamie (sip)
- Joe Tantillo (sip)
- Khyati (sip)
- Matt Nurte (sip)
- Rodina (sip)
- Shannon\_Burke (sip)
- Stormcloud (sip)
- Stormcloud Brew (sip)
- Tommy Carr 32 (sip)
- VoIP Master (sip)
- WestBentivoliPHS (sip)

Enter peering details as outlined below then press Submit and Apply..

VOICEMAIL SETTINGS

Applications

- Asterisk API
- Conferences
- Directories
- Languages
- Misc Applications
- Misc Destinations
- Music on Hold
- PN Sets
- Paging and Intercom
- Parking Lot
- Phonebook
- Scheduled Announcements
- System Recordings
- VoiceMail Blasting
- Wake Up Calls

Remote Access

- Callback
- DISA

**Outgoing Settings**

Trunk Name: IPOffice

PEER Details:

```
host=192.168.1.250
port=5050
type=friend
quality=yes
permit=192.168.1.0/255.255.255.0
insecure-port,invite
trunk=yes
context-from-internal
```

Incoming Settings

USER Context:

USER Details:

Registration

Register String:

Submit Changes Duplicate Trunk

After your dial plan for you outbound and inbounds routes are defined you should be able to talk between the 2 systems now.