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SIP Trunking Configuration Guide
for
ESI Communications Server 50/100
12.05.49.1 Software Version
ESI System Programmer 1.2.14.19

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Business®

1 Audience

This document is intended for the SIP trunk customer's technical staff and Value Added Retailer (VAR) having installation and operational responsibilities.

2 Introduction

This Configuration Guide describes configuration steps for Cox SIP trunking to an ESI Communications Server 50/100 (ESI 50) IP-PBX. Cox SIP trunking is a scalable and efficient IP trunking telecommunication solution for your business that provides all the traditional services such as Direct Inward Dialing, Hunting, Calling Name, Calling Number, Local/Long Distance and Business Continuity options, including:

- Burstable Trunk Capacity – Dynamically increases call capacity during peak busy periods so your customers never receive a busy signal.
- Call Forward Always – On the trunk group pilot number for all calls in case of an outage (i.e., flood, fire, loss of power, etc.).
- Call Forward Not Reachable – On the trunk group pilot number that operates on a per-call contingency basis to forward the call to any PSTN number (i.e., call center or alternate office location) during temporary call completion impairments.
- Route Exhaustion – Automatic reroute of trunk group calls to any PSTN phone number (i.e., a call center) if calls can't be completed to the PBX.
- Support for geo-redundant PBX deployments and automatic reroute of SIP trunks to the backup customer data center.

All calls are routed over Cox's national fiber network with guaranteed Quality of Service (QoS); <u>calls never traverse</u> the Internet.

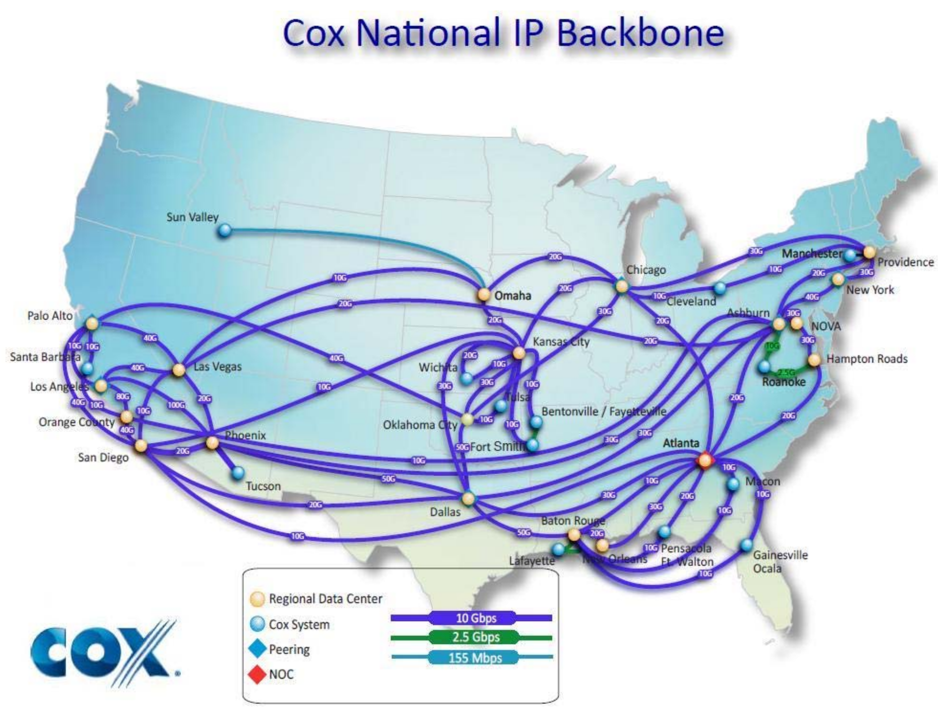


Figure 1 - Cox Fiber Network

2.1 tekVizion Labs

tekVizion Labs™ is an independent testing and Verification facility offered by tekVizion PVS, Inc. ("tekVizion"). tekVizion Labs offers several types of testing services including:

- Remote Testing – provides secure, remote access to certain products in tekVizion Labs for pre-Verification and ad hoc testing
- Verification Testing – Verification of interoperability performed on-site at tekVizion Labs between two products or in a multi-vendor configuration ("solution Verification")
- Product Assessment – independent assessment and verification of product functionality, interface usability, assessment of differentiating features as well as suggestions for added functionality, stress and performance testing, etc.

tekVizion is a systems integrator specifically dedicated to the telecommunications industry. Our core services include consulting/solution design, interoperability/Verification testing, integration, custom software development and solution support services. Our services helps service providers achieve a smooth transition to packet-voice networks, speeding delivery of integrated services. While we have expertise covering a wide range of technologies, we have extensive experience surrounding our FastForward>> practice areas which include: SIP Trunking, Packet Voice, Service Delivery, and Integrated Services.

The tekVizion team brings together experience from the leading service providers and vendors in telecom. Our unique expertise includes legacy switching services and platforms, and unparalleled product knowledge, interoperability and integration experience on a vast array of VoIP and other next-generation products. We rely on this combined experience to do what we do best: help our clients advance the rollout of services that excite customers and result in new revenues for the bottom line. tekVizion leverages this real-world, multi-vendor integration and test experience and proven processes to offer services to vendors, network operators, enhanced service providers, large enterprises and other professional services firms. tekVizion's headquarters, along with a state-of-the-art test lab and Executive Briefing Center, is located in the Telecom Corridor® in Richardson, Texas.
(*For more information on tekVizion and its practice areas, please visit tekVizion Labs's web site at www.tekVizionlabs.com.)*

3 SIP Trunking Network Components

The network for the SIP trunk reference configuration is illustrated below and is representative of an ESI Communications Server 50 (ESI 50) configuration.

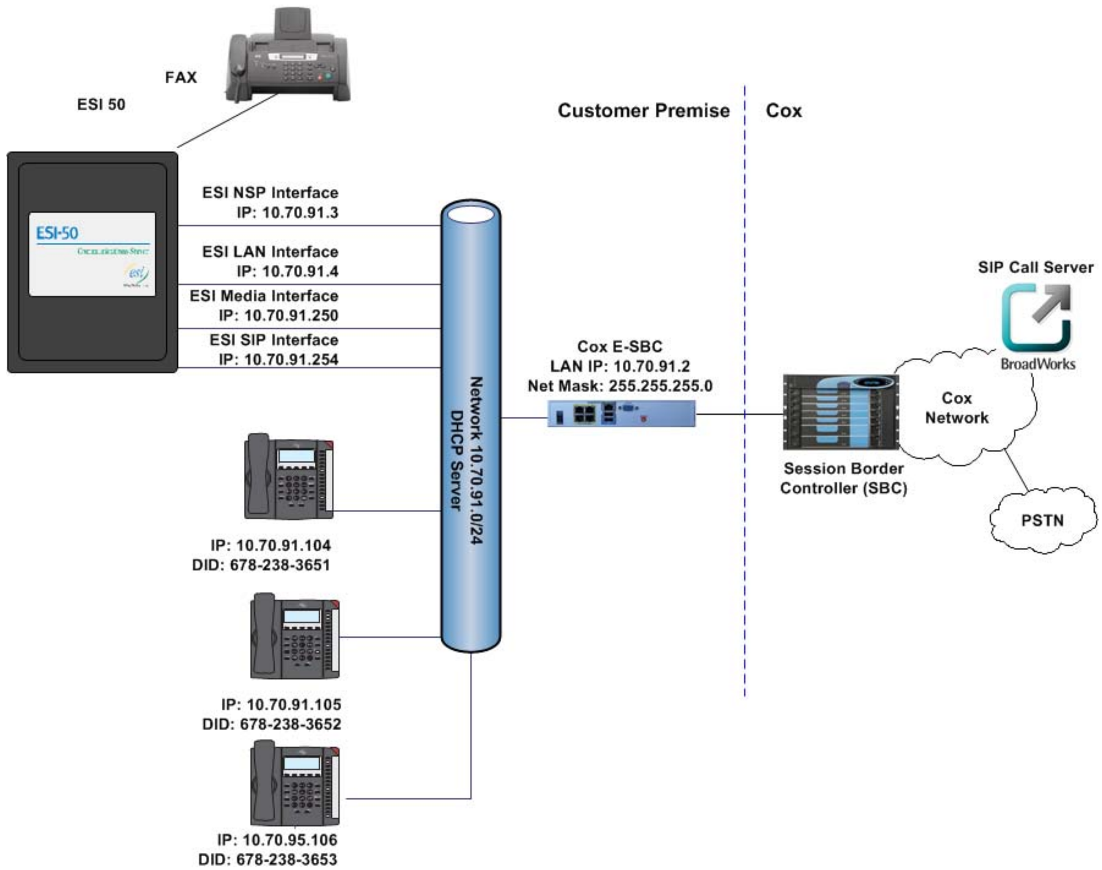


Figure 2 - SIP Trunk Lab Reference Network

Note: The ESI 50 does not offer DHCP server for dynamic IP address assignment for the ESI phones; however, the Cox Enterprise Session Border Controller (E-SBC) requires a static LAN IP address that must be manually assigned by the LAN network administrator. The DHCP server is provisioned on the Ethernet switch. The DHCP's IP address pool is constrained so that the E-SBC can be assigned an IP address outside of the pool.

The lab network consists of the following components:

- ESI 50 IP-PBX for voice features, SIP proxy and SIP trunk termination.
- Various SIP phones on the local LAN.
- The Cox E-SBC is the Edgewater Networks (www.edgewaternetworks.com) EdgeMarc appliance. The EdgeMarc is the service demarcation point between customer's LAN network and Cox's WAN network and provides firewall/NAT traversal, telemetry and SIP Application-level gateway. The EdgeMarc has diverse routes to a primary and secondary Acme SBC.
- Acme Packet Net-Net 9200 Session Border Controllers (SBC).

3.1 Hardware Components

- ESI Communications Server 50
- Analog fax machine
- EdgeMarc 4550 E-SBC

3.2 Software Requirements

- ESI Communications Server 50 12.05.49.1
- ESI System Programmer 1.2.14.19
- EdgeMarc 4550 9.12.5 Release

4 Features

4.1 SIP Registration Method

Cox Network requires SIP REGISTER support to allow the IP-PBX to originate calls from the IP-PBX and to send calls to the PBX from the PSTN. The ESI 50 supports SIP Register with authentication. Cox will provide the SIP trunk authentication credentials to the customer, which must be provisioned in the ESI 50. How to configure these in the ESI 50 are shown in Section 6.3.2.

4.2 Features Supported

- Basic calls using G.711ulaw
- Calling Party Number Presentation
- Call Transfer
- Call Forwarding
- Call Hold and Resume
- Call Pickup
- Call Waiting
- DND
- Hunt groups (Simultaneous and Sequential Ring)
- Three-Way Calling
- PBX Auto Attendant to Off-net Numbers
- PBX Account Codes
- G711 Fax
- Dial-Up Modem
- E911 Call
- RFC2833 transcoding
- PBX-Defined Caller ID (spoofing)

4.3 Features Not Supported

- Anonymous call
- Call Park
- PBX Authorization Codes
- T.38 FAX

5 Caveats and Limitations

- The ESI software version must be 12.5.49.1 or greater with ESI System Programmer 1.2.14.19 or greater that provides the pre-loaded Cox ITSP profile.
- ESI 50 hairpins both call legs during call transfer and call forward to PSTN, meaning the SIP sessions are not released after transfer. The sessions are released when the calls are released.
- Anonymous Calling is NOT Supported. However a workaround can be Section 6.3.5.
- Call Park is NOT supported. System Hold can be used in its place.
- PBX Authorization Codes are NOT supported.

6 Configuration

6.1 Configuration Checklist

In this section we present an overview of the steps that are required to configure ESI 50 for SIP Trunking as well as all features that were tested.

If the ESI PBX is already operational, please skip to Step 3 to begin the SIP Trunk configuration.

Table 1 – PBX Configuration Steps

Step	Description	Reference
Step 1	NSP IP Address	Section 6.3.1.1
Step 2	Base IP Address	Section 6.3.1.2
Step 3	Signaling/Media IP Address	Section 6.3.1.3
Step 4	Sip Account Programming	Section 6.3.4
Step 5	Sip Trunk Programming	Section 6.3.5
Step 7	Anonymous Call	Section 6.3.7
Step 9	Incoming Routing	Section 6.4.1
Step 10	Enable Incoming Caller ID	Section 6.4.2
Step 11	Outgoing Dial Plan	Section 6.4.3
Step 13	Send CLI Number	Section 6.4.4
Step 14	Calling Line ID Spoofing	Section 6.4.5

6.2 IP Address Worksheet

The specific values listed in the table below and in subsequent sections are used in the lab configuration described in this document, and are for **illustrative purposes only**. The customer must obtain and use the values for your deployment.

Table 2 – IP Addresses

Component	Cox Lab Value	Customer Value
EdgeMarc E-SBC		
• LAN IP Address	10.70.91.2	
• LAN Subnet Mask	255.255.255.0	
ESI 50 IP PBX		
• ESI SIP Signaling Interface The Signaling interface is on the same subnet as the LAN IP Address of the E-SBC. If this is not the case, then Layer 3 routing must be in place.	10.70.91.254	
• ESI NSP Remote Support Interface The NSP Remote Support interface is used to gain access to the IP-PBX	10.70.91.3	
• ESI Media Interface	10.70.91.250	
• ESI LAN This the IP address that the phones communicate with.	10.70.91.4	
• Default Gateway The Default Gateway must be the LAN Network default Gateway. This will allow the administrator to log in via his\her workstation if the workstation is on a different network	10.70.91.1	

6.3 ESI 50 Detailed Configuration Steps

Equipment used for configuration setup:

- ESI 50 software version release 12.05.44.0
- ESI System Programmer Version 1.2.14.17

6.3.1 System IP Addresses

The ESI Communications Server 50 (ESI50) has four IP addresses that make up the ESI50 LAN network. Please reference **Figure 2** and **Table 2** for the network topology in this example configuration. Please use the actual IP addresses for your network.

- ESI NSP IP address is defined in F824. This is the IP address that the ESI System Programmer communicates with.
- ESI Base IP Address is defined in F821 and F822. ESI Base IP address is used by the phones to communicate with ESI50.
- ESI Signaling Port IP address is defined in F84. This is the IP address that the E-SBC uses for SIP Trunking Devices and the Default Dial Rules.
- ESI Media Port IP address is defined in F84. This is the IP address that is used for media (RTP).

6.3.1.1 ESI NSP IP Address

1. Navigate to **F8 IP Programming > F82 Local Programming > F824 NSP IP Programming**
2. Set **NSP Private IP Address**: For this example 10.70.91.3 is used. This is the ip address the ESI Gui connects to.
3. Set **NSP Subnet Mask:** For this example 255.255.255.0 is used.
4. Set the **NSP Gateway IP Address:** For this example 10.70.91.1 is used. Plese confirm network address to be used with your network administrator.
5. Click on **Save** if IP address is modified.

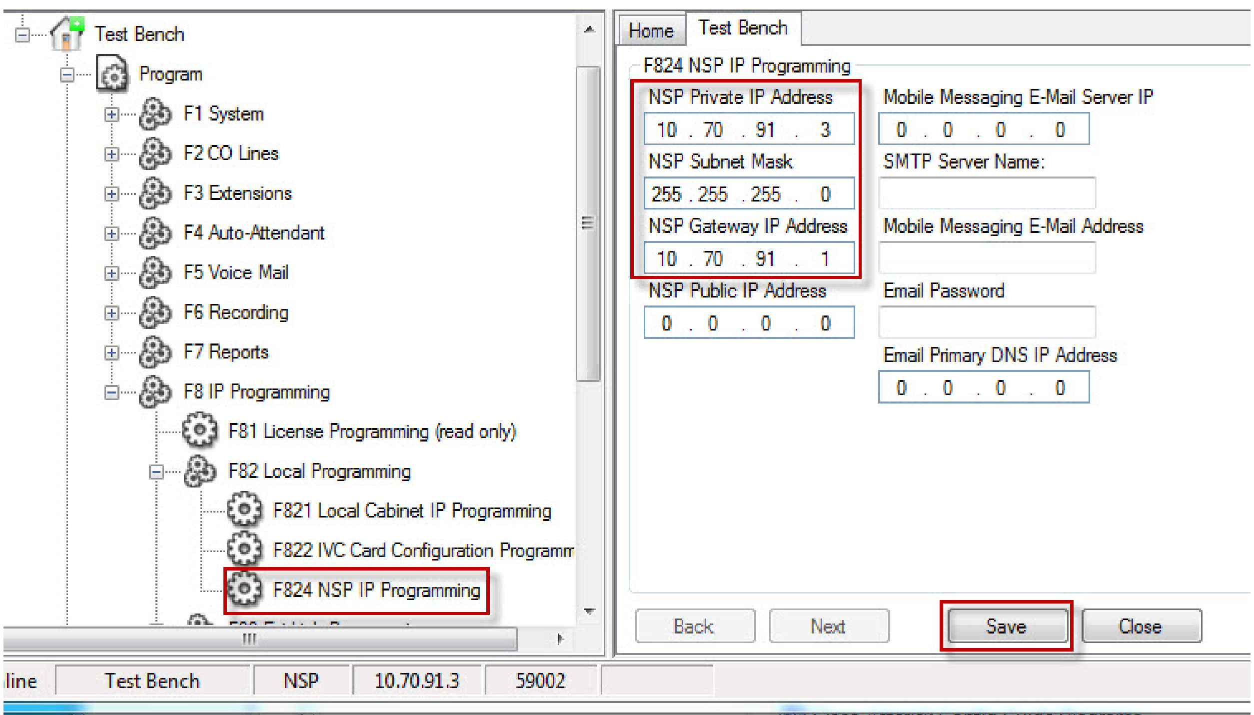


Figure 3 ESI IP Address

6.3.1.2 **ESI Base IP Address**

1. Navigate to **F8 IP Programming > F82 Local Programming > F821 Local Cabinet IP Programming**
2. Set Base **IP Address**: For this example 10.70.91.4 is used. This is the ip address the phones use for communication to the ESI50
3. Set Subnet **Mask:** For this example 255.255.255.0 is used.
4. Set the **Gateway IP Address:** For this example 10.70.91.1 is used. Plese confirm network address to be used with your network administrator.
5. Click on **Save** if IP address is modified.

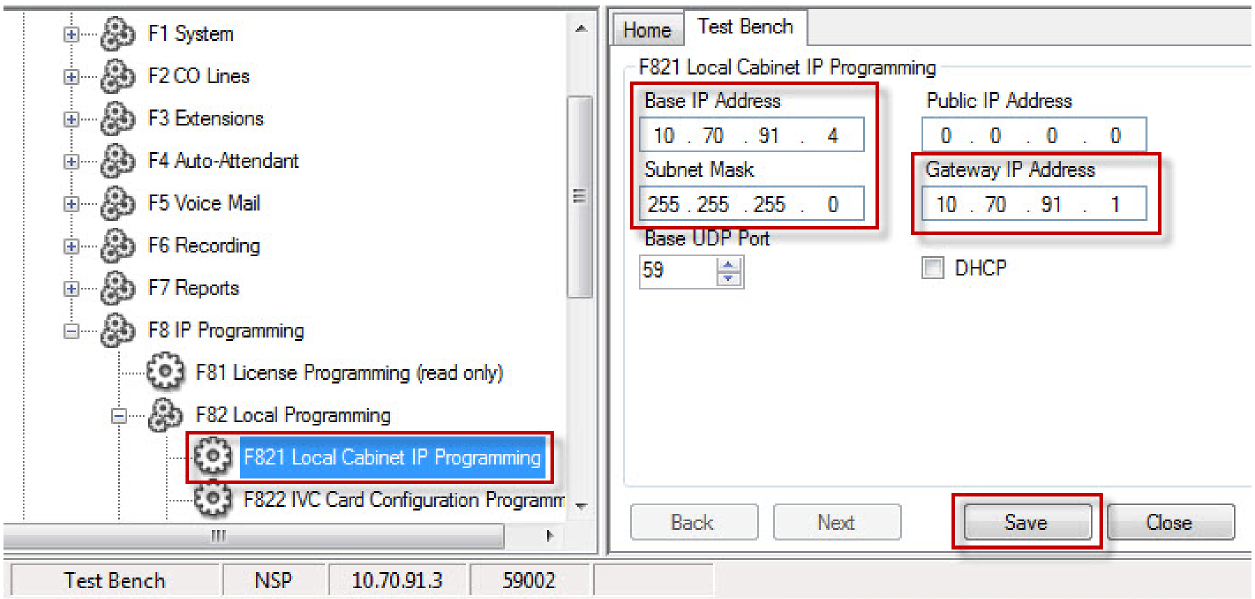


Figure 4 Base IP Address

6. Navigate to **F8 IP Programming > F82 Local Programming > F822 IVC Card Configuration Programming**
7. Set **IP Address:** For this example the value defined in Step 2 above is used.
8. Set **Subnet mask:** 255.255.255.0.
9. Confirm **UDP Port:** 59101
10. Set **Gateway IP:** For this example the value defined in Step 4 above is used.
11. Set **Vlan Cos:** 1
12. Check **Master**

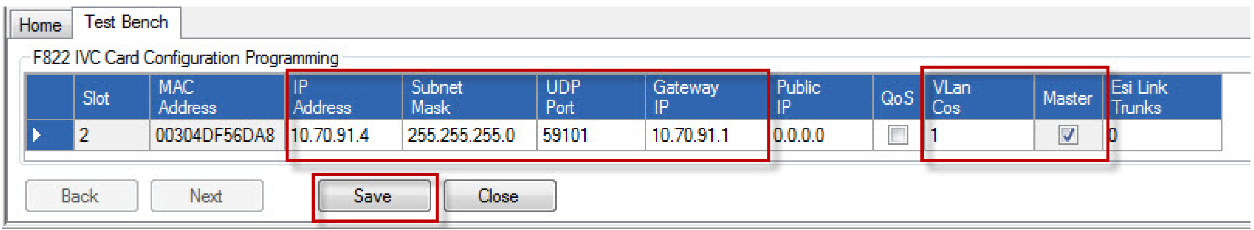


Figure 5 ESI IVC Card IP Address

6.3.1.3 ESI Signaling/Media IP Address

1. Navigate to **F8 IP Programming > F84 SIP Card Configuration Programming**
2. Record the information in **Card:** This will be used in Section 6.3.2 step 4
3. Set **Card Type:** SIGNALING
4. Set **Primary Processor IP:** This is the IP address that the E-SBC uses for SIP Trunking Devices and the Default Dial Rules. Please reference **Figure 2** and **Table 2** for the network topology that is tested in this example configuration. For this example 10.70.91.254 is used.
5. Set **Subnet Mask:** For this example 255.255.255.0 is used.
6. Set the **Gateway IP Address:** For this example 10.70.91.1 is used. Plese confirm network address to be used with your network administrator.
7. Set **Public IP:** This is the static LAN IP address of the Cox E-SBC. Please use the actual E-SBC LAN IP for your network. The IP Address used in this configuration is 10.70.91.2. The E-SBC LAN IP address may/will be different from this example. Please see **Figure 2** and **Table 2** for the IP address scheme.
8. Set **Secondary Processor IP:** This is the IP address for RTP. For this example 10.70.91.250 is used.

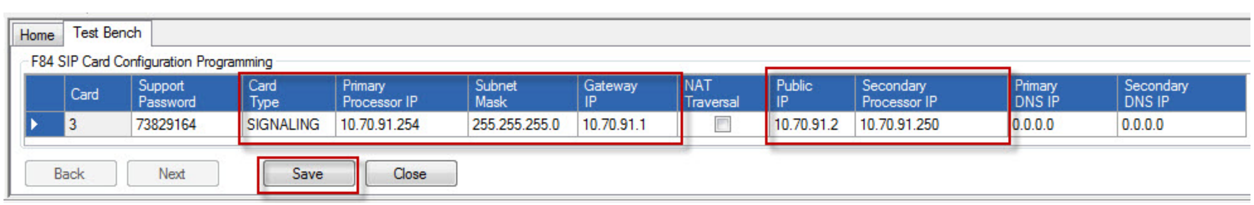
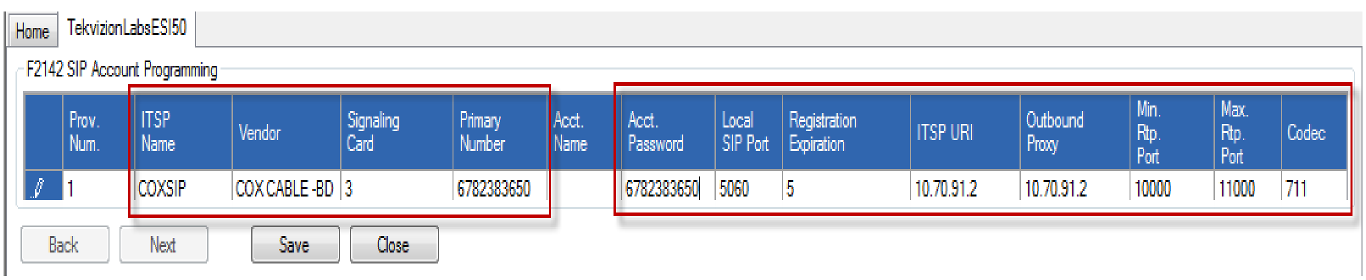


Figure 6 Signaling/Media IP Address

6.3.2 SIP Account Programming

1. Navigate to **F2 CO Lines > F2142 SIP Account Programming**
2. Set **ITSP Name**:This field is a descriptive name that will be used in Section 6.3.3. For this example COXSIP is used.
3. Set **Vendor:** Select COX CABLE-BD from the list.
4. Set **Signaling Card:** This is the card number defined in Section 6.3.1.3
5. Set **Primary Number:**This is the Trunk Group Pilot Number. For this example 6782383650 is used.
6. Set **Acct.Passsword:**This is the Registration Password that will be provided by your Cox Account Representative. For this example 6782383650 is used.
7. Set **Local SIP Port:**5060
8. Set **Registration Expiration:** 120
9. Set **ITSP URI:** This is the static LAN IP address of the Cox E-SBC. Please use the actual E-SBC LAN IP for your network. The IP Address used in this configuration is 10.70.91.2. The E-SBC LAN IP address may/will be different from this example. Please see **Figure 2** and **Table 2** for the IP address scheme.
10. Set **Outbound Proxy:** This is the static LAN IP address of the Cox E-SBC. Please use the actual E-SBC LAN IP for your network. The IP Address used in this configuration is 10.70.91.2. The E-SBC LAN IP address may/will be different from this example. Please see **Figure 2** and **Table 2** for the IP address scheme.
11. Set **Min. Rtp. Port:** This field is used to set the minimum value of the lowest number RTP port. For purpose of this example 10000 is used.
12. Set **Max. Rtp. Port:** This field is used to set the maximum value of the RRP port. For purpose of this example 11000 is used.
13. Set **Codec:** 711
14. Click **Save**

The actual SIP Registration Username and Password will be provided by your Cox Account Representative and must be kept confidential! The Trunk Group Pilot Number (username) is used here for illustration purposes only!



	Prov. Num.	ITSP Name	Vendor	Signaling Card	Primary Number	Acct. Name	Acct. Password	Local SIP Port	Registration Expiration	ITSP URI	Outbound Proxy	Min. Rtp. Port	Max. Rtp. Port	Codec
	1	COXSIP	COX CABLE -BD	3	6782383650		6782383650	5060	5	10.70.91.2	10.70.91.2	10000	11000	711

Figure 7 SIP Account Programming

6.3.3 Sip Trunk Programming

1. Navigate to **F2 CO Lines > F214 SIP Line Programming > F2141 SIP Trunk Programming**
2. Set **Name**: For this example COXTRUNK is used.
3. Set **Outbound Group**: For this example 9 is used. Right click on the field to choose.
4. Set **Ring Tone**: For this example 6 is chosen. Right click on the field to choose.
5. Set ITSP Name: This is the name that is defined in Section 6.3.2 Step 2.
6. Set **Ring 1 (Day):** ID 1
7. Repeat Steps 2 – 5 for each trunk member that needs to be programmed.

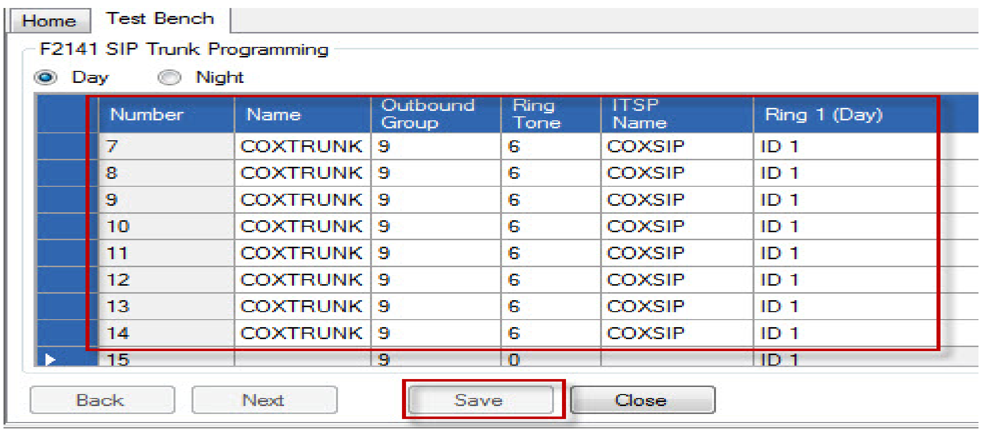


Figure 8 SIP Trunk Programming

8. Navigate to **F2 CO Lines > F214 SIP Line Programming > F2145 Piolt Table forSIP Trunks**
9. Set to **Enable**

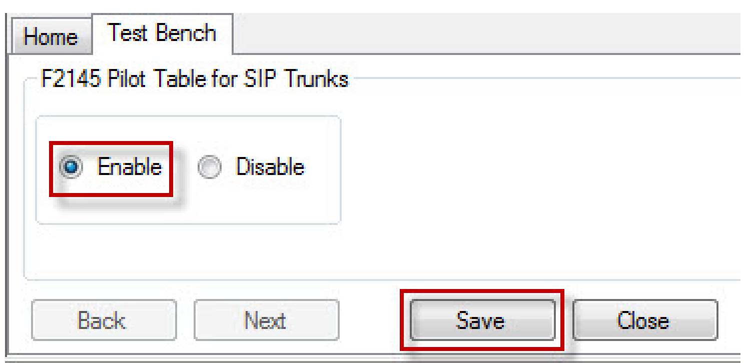


Figure 9 Pilot Table for SIP Trunks

10. Navigate to **F2 CO Lines > F22 CO Access Deny Tables > F225 PRI Pilot Number**
11. Set **DID/Pilot:** Enable
12. Set **Number**:In this example 6782383650 is used. This is the Pilot Number of the trunk group that you have been assigned by the COX Representative.
13. Set **MAX Channesls**: In this example 10 is used. This is the maximum number of concurrent calls.
14. Set **Ring Tone:** For this example 0 is used.
15. Set **Ring 1 (Day):** For this example all incoming calls that are not defined in **Figure 11** will terminate to ID1 which is the AA.

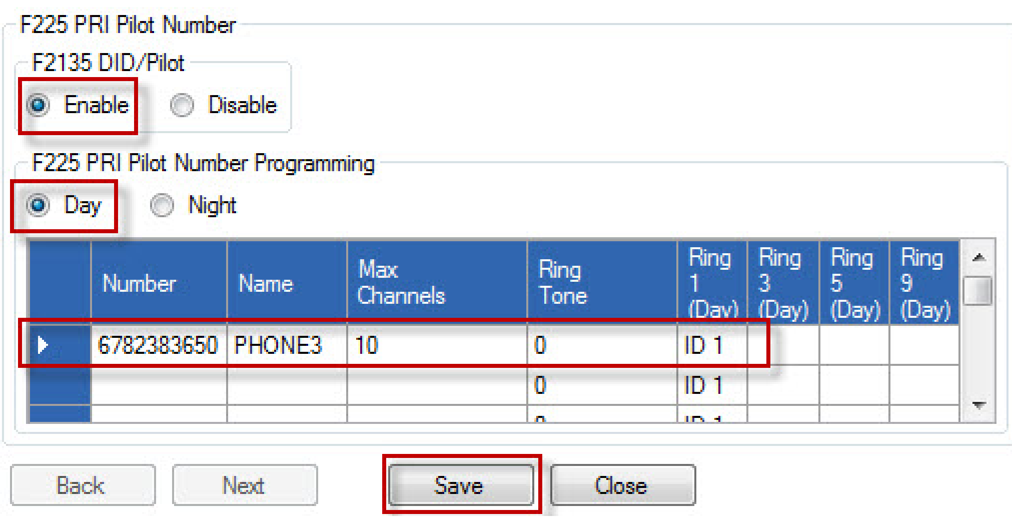


Figure 10 F225 Pilot Number

6.3.4 Anonymous Call

The ESI50 does not support Anonymous Call, however it is possible to configure the Caller ID with 000 000 0000 so that the terminating number does not see the Caller's telephone number. This procedure is covered in Section 6.4.5.

6.4 ESI Dial Plan Configuration

For purpose of this example ESI50 is configured to dial 9 which is the access code for the SIP trunk. The steps below give detailed step by step instructions for Inbound and out bound calls.

6.4.1 Incoming Routing

To provision DID numbers in ESI50 is a very simple process. The steps below define this process. The DID numbers that are used below are test numbers. Use only the range that is assigned to you from your Cox representative

1. Navigate to **F2 CO Lines > F22 CO Access Deny Tables > F224 DID & DNIS/ANI Transfer**
2. Each Entry will be associated with a extension, Dept or ID.
3. The first entry in **Figure 11** is Exception. This cannot be changed and directs the call to the default station if the DID is not provisioned.

4. Click Add
5. Set "**Number**": 6782383651 to the right under "Call Fwd (Day) and Call Fwd (Night)" Right-Click and choose extension 112.
6. Repeat Step 4 and 5 for each entry in **Figure 11**.
    - When 6782383651 is received the call terminates to extension 112
    - When 6782383652 is received the call terminates to extension 113
    - When 6782383653 is received the call terminates to extension 114
    - When 6782383654 is received the call terminates to AA ID 1 Error! Reference source not found.
    - When 6782383655 is received the call terminates to Dept 280 Error! Reference source not found.
    - When 6782383656 is received the call terminates to Dept 281 Error! Reference source not found.
    - When 6782383657 is received the call terminates to extension 108
7. Click **Save**

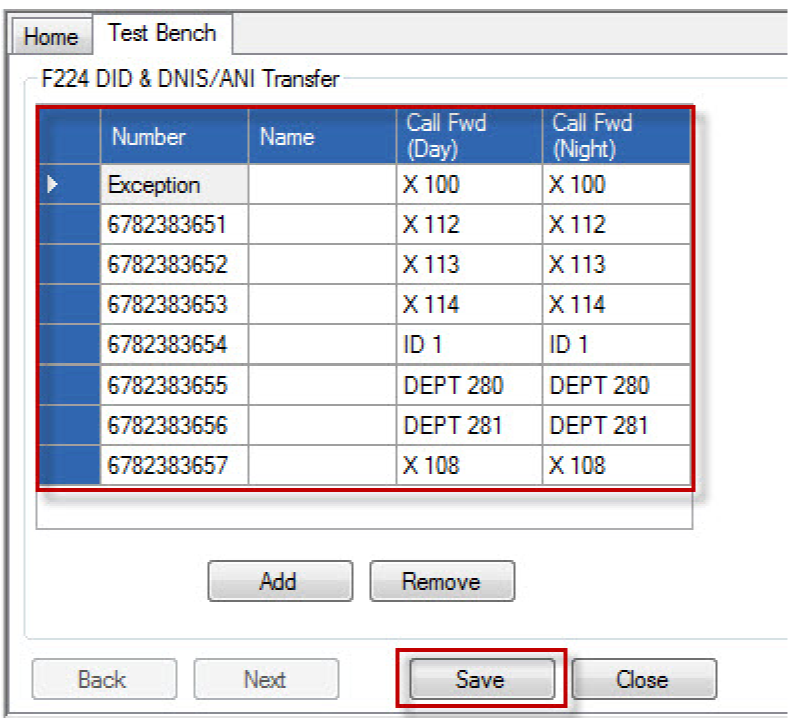


Figure 11 Incoming DID Termination

6.4.2 Enable Incoming Caller ID

1. Navigate to **F2 CO Lines > F24 Caller ID**
2. Set **Caller ID:** Enabled

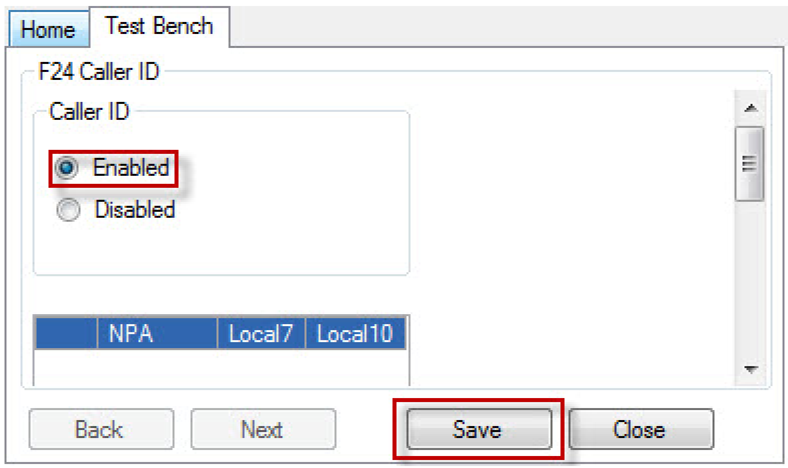


Figure 12 Caller ID Enable

6.4.3 Outgoing Dial Plan

In Section 6.3.3 the trunk members are assigned to Outbound Group 9. That which is not explicitly denied in F222 Toll Restrictions Deny table is allowed.

1. Navigate to **F3 Extensions > F31 Definition and Routing**
2. Scroll down until the „"**Extension**" column shows the extension being configured. In this example extension 112 is being configured.
3. Set **Extension** 112 **Line Groups:** 9 The Line Group value is set by Right-Click and choose the access code that the extension is allowed to dial. In this example Line Group 9 is used to give the programmed extension access to the trunk group.

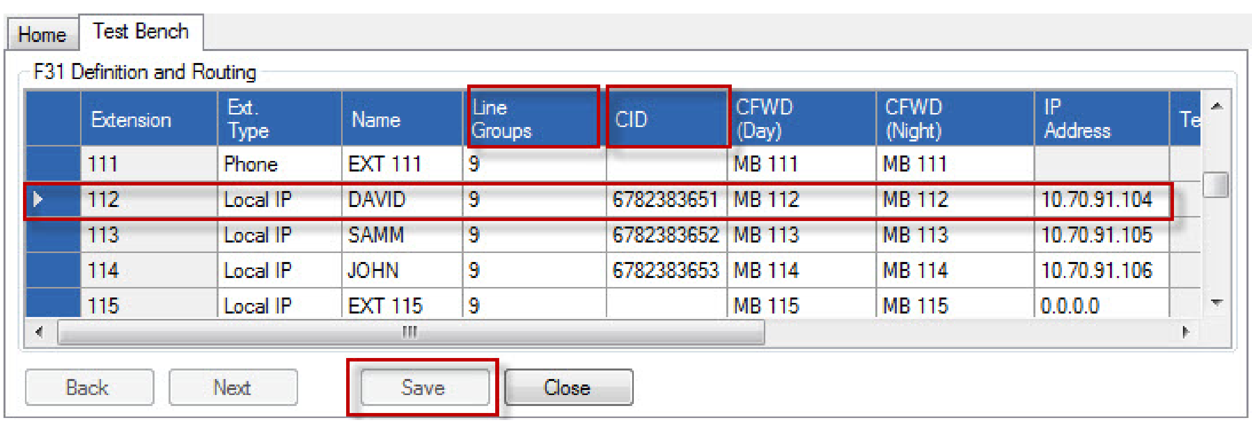


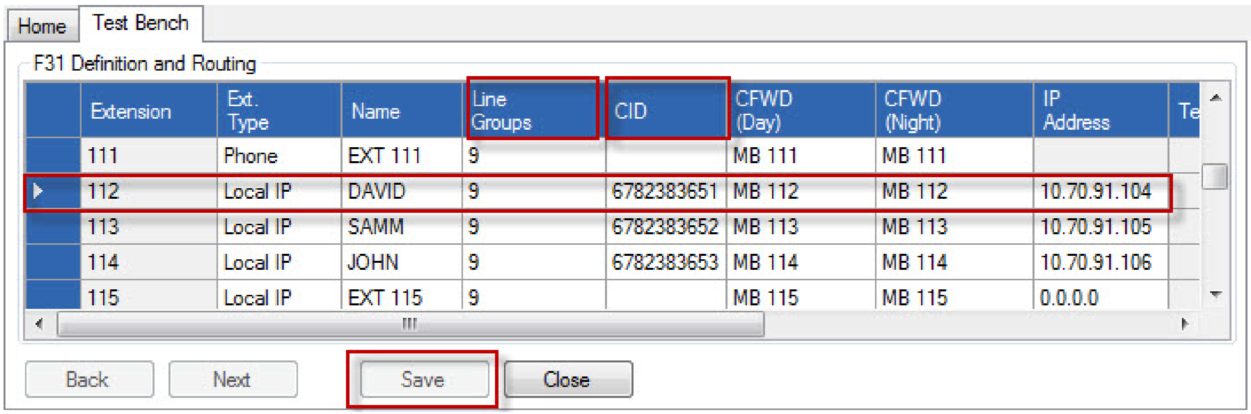
Figure 13 Line Group and Caller ID

6.4.4 Send CLI Number

The Calling Line ID (CLID) is configured to send phone user's 10-digit DID number.

Navigate to **F3 Extensions > F31 Definition and Routing**

1. Scroll down until the "**Extension**" column shows the extension being configured. In this example extension 112 is being configured.
2. Set **Extension** 112 **CID:** 6782383651. The CID value is set by typing the correct value in the field. In this example 6782383651 is used for extension 112 CID.
3. Click **Save**



	Extension	Ext. Type	Name	Line Groups	CID	CFWD (Day)	CFWD (Night)	IP Address	Te
	111	Phone	EXT 111	9		MB 111	MB 111		
▸	112	Local IP	DAVID	9	6782383651	MB 112	MB 112	10.70.91.104	
	113	Local IP	SAMM	9	6782383652	MB 113	MB 113	10.70.91.105	
	114	Local IP	JOHN	9	6782383653	MB 114	MB 114	10.70.91.106	
	115	Local IP	EXT 115	9		MB 115	MB 115	0.0.0.0	

Figure 14 F31 CLID

6.4.5 Calling Line ID Spoofing

Calling ID Spoofing is the practice of sending a CLID that is not within the Cox assigned DID telephone number range(s). A legitimate use of CLID spoofing is the PBX customer wants to display their toll free number on non-emergency PBX originated calls so the dialed party can call the company back. To enable CLID spoofing support, the PBX customer must:

1. Contact your Cox sales representative to request spoofed CLID support on the Cox network, otherwise Cox will block calls that present a spoofed CLID.
2. For this example, we will use Extension 112 and assign 8006781234.
3. Enter the toll free or other telephone number instead of the 10-digit DID number to be displayed for each extension as described in Section 6.4.5 Steps 2 and 3 above.