

XO SIP Service

Customer Configuration Guide for Siemens
HiPath Open Office ME/OpenScope Office MX V2
with XO SIP



1. Overview

About This Document

This document describes interoperability between XO SIP Package 1 (G.711) and Package 2 (G.729a) and the Siemens HiPath Open Office ME/OpenScope Office MX V2 unified communications system with software version HOO_V1_R4.6.0_102, deployed with an XO-provided Cisco 2432 IAD as the router/demarcation device.

This document assumes the audience has a general understanding of network provisioning and the connectivity requirements of XO Communications SIP service offering.

Known Issues

While XO certifies interoperability between XO SIP service and the IP PBX as outlined herein, the following known issues were identified during Interoperability testing. The customer should be aware that certain features and functions may not be fully supportable. In some cases, special configurations and/or PBX software patches may be available from the vendor.

- 1) XO SIP optional Call Center feature is not fully supported with the Siemens OpenScope Office MX V2.
- 2) Inbound CNAM (caller ID name) is not displayed on extensions, as the system does not support this feature on SIP trunks. It does display inbound caller ID number, however this number is displayed on the user's extension screen with prefix. The prefix is digit used to grab trunk to make outbound call. This is default behavior of the Siemens OpenScope Office MX V2.
- 3) Codec preference needs to be configured on individual IP Phone extensions (using web interface of phone) as system's codec preference doesn't apply to IP Phone extensions. It only applies to analog extensions.
- 4) Siemens OpenScope Office MX V2 does not support in-band RTP DTMF with IP phones; it only supports DTMF as RFC2833 with IP phones. Both types of DTMFs are supported with analog extensions.
- 5) Siemens OpenScope Office MX V2 requires that an account number is configured for the SIP trunk. For testing purposes, one of DIDs was configured as the account number. That DID was not used with any other extension.

Registration Method

Static registration is utilized between the Siemens OpenScope Office MX V2 IP PBX and the XO call agent.

XO SIP Service Packages Supported

Pkg	Codec	DTMF	Fax
1	G.711	RFC2833 is required for DTMF	G.711 and T.38
2	G.729a	RFC2833 is required for DTMF	G.711 and T.38

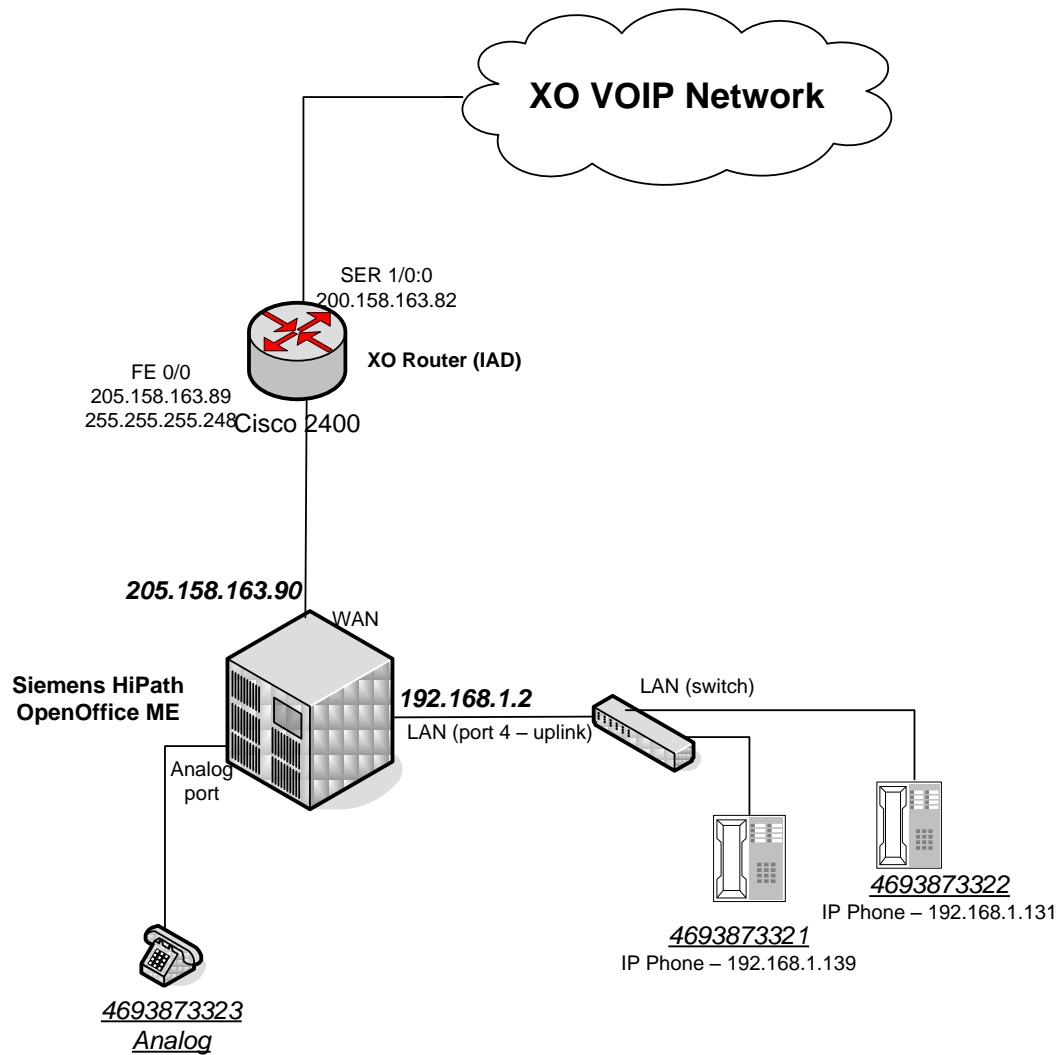
2. Testing of Siemens OpenScape Office MX V2 IP PBX

2.1. Software and Hardware Versions Tested

1. Cisco 2400 used as a router
Cisco IOS Software, 2400 Software (C2430-IS-M), Experimental Version
12.4(20060718:201451) [acphan-V124_4_T2_1 104]
System image file is "flash:c2430-is-mz.xo"
2. Siemens HiPath OpenOffice ME (IP PBX has been subsequently re-branded as OpenScape Office MX V2)
Software version: HOO_V1_R4.6.0_102
3. IP Phones
Model no.: OptiPoint 430 Standard
4. Fax machines:
G3: SHARP UX-B20
SG3: Brother - IntelliFax 1860c

2.2. Lab Test Configuration

The following diagram shows the configuration used during lab testing.



Note: Above lab setup only shows main lab network elements.

- In this setup no NAT were configured on XO Router (IAD).

3. Siemens OpenScape Office MX V2 Configuration

In This Section XO performed the minimum amount of configuration required to achieve successful completion of test calls over XO SIP. It is beyond the scope of this document and the testing efforts to show a complete configuration, therefore screenshots of the GUI interface are provided only for the details of the SIP trunk configuration that are relevant to interfacing with XO’s SIP product.

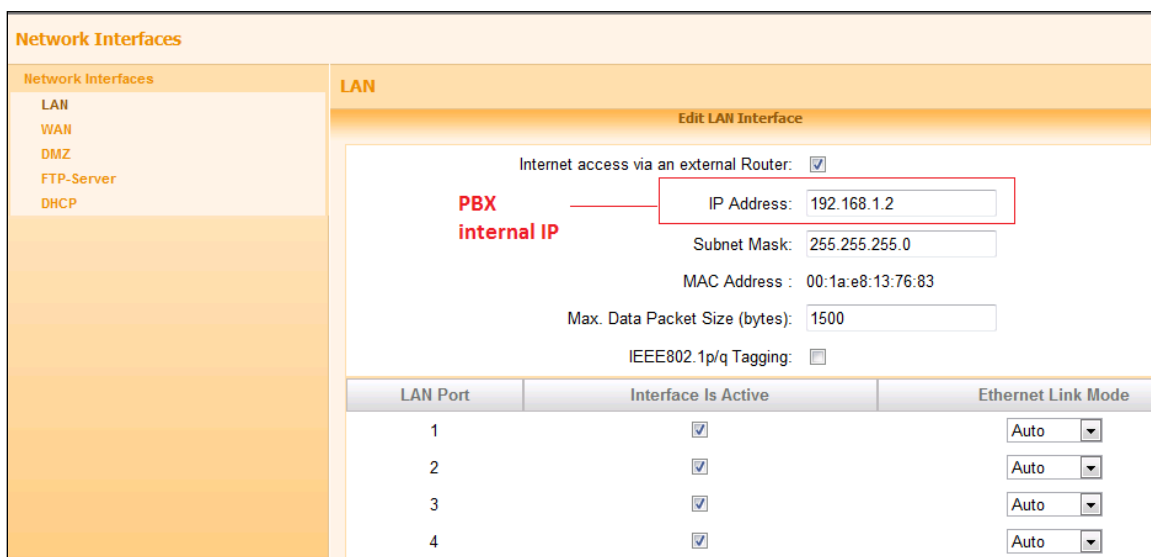
This section contains screenshots which detail the SIP trunk configurations.

Basic configuration screen shots of Siemens HiPath OpenOffice ME/ OpenScape Office MX V2 for XO SIP interoperability testing.

Expert mode >> Telephony Server >> Network Interface >> LAN

“IP Address” = PBX internal (private) IP Address

“Subnet Mask” = PBX internal (private) subnet mask



- “Internet access via an external router” was selected under LAN for lab setup.

Expert mode >> Telephony Server >> Network Interface >> WAN

“IP Address” = PBX external (public) IP Address

“Subnet Mask” = PBX external (public) subnet mask

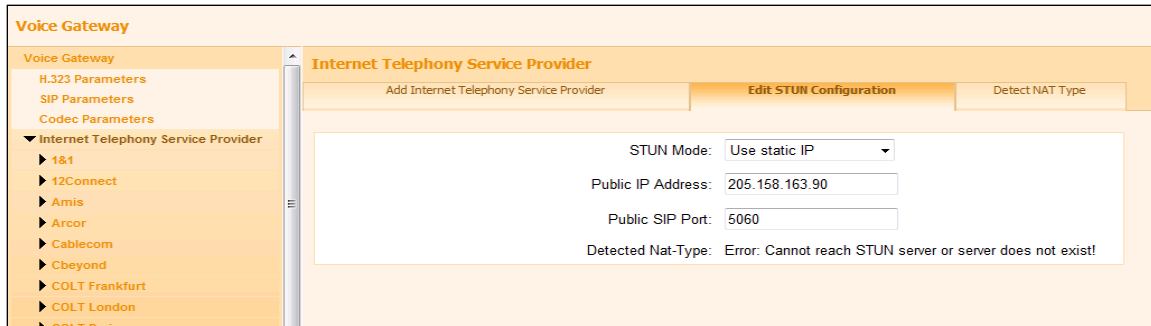
The screenshot displays the 'Network Interfaces' configuration window, specifically the 'WAN' tab. On the left, a sidebar lists 'LAN', 'WAN', 'DMZ', 'FTP-Server', and 'DHCP'. The main area is titled 'WAN' and contains several sections:

- Internet Service Provider Selection:** A dropdown menu set to 'LAN Connection Type TCP/IP'.
- Internet access via an external Router:** An unchecked checkbox.
- Automatic Address Configuration (via DHCP):** An unchecked checkbox.
- IP Address:** A text field containing '205.158.163.90', highlighted with a red box. A red arrow points to this field from the text 'PBX external IP'.
- Subnet Mask:** A text field containing '255.255.255.248'.
- MAC Address:** A text field containing '00:1a:e8:13:76:82'.
- Ethernet Link Mode:** A dropdown menu set to 'Auto'.
- Max. Data Packet Size (bytes):** A text field containing '1500'.
- Network Address Translation:** An unchecked checkbox.
- Bandwidth Control for Voice Connections:** A dropdown menu set to 'None'.
- Bandwidth for Downloads:** A text field containing '10000'.
- Bandwidth for Uploads:** A text field containing '10000'.
- Bandwidth Used for Voice/Fax (%):** A text field containing '80'.
- IEEE802.1p/q Tagging:** An unchecked checkbox.
- IEEE802.1x/a VLAN ID:** A text field containing '0'.

At the bottom of the configuration area, there are three buttons: 'Apply', 'Undo', and 'Help'.

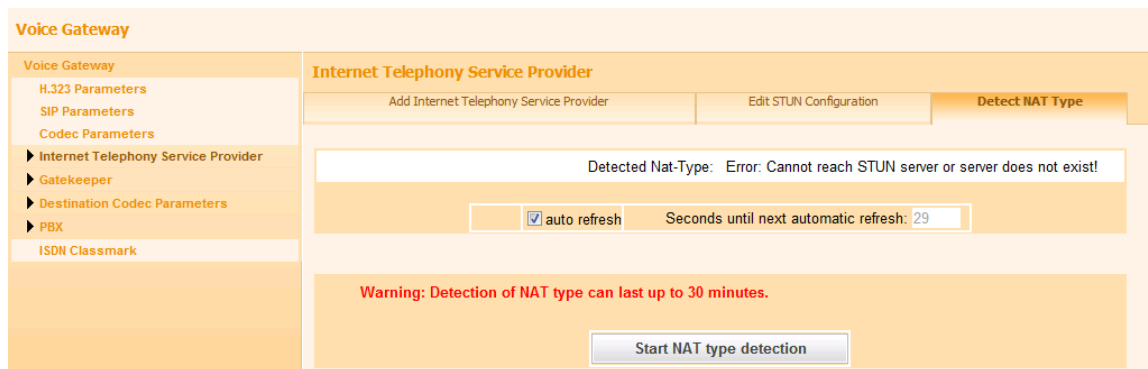
- “Internet access via an external router” was not selected under WAN for lab setup.

Expert mode >> Voice Gateway >> Internet Telephony Service Provider >> STUN configuration.



- *STUN Mode was setup as “Use Static IP” with “public IP Address” = IPPBX external (public) IP. This was required with lab setup.*

Expert mode >> Voice Gateway >> Internet Telephony Service Provider >> NAT configuration.



- *NAT was not enabled on IPPBX or XO router with lab setup.*

Expert mode >> Telephony Server >> Network Interface >> DMZ

Network Interfaces	
<p>Network Interfaces</p> <ul style="list-style-type: none"> LAN WAN DMZ FTP-Server DHCP 	<p>DMZ</p> <p style="text-align: center;">Edit DMZ Interface</p> <p>Interface Is Active: <input type="checkbox"/></p> <p>Automatic Address Configuration (via DHCP): <input type="checkbox"/></p> <p>Accept IP Address of the Default Router: <input type="checkbox"/></p> <p>Accept IP Address of the DNS Server: <input type="checkbox"/></p> <p>Accept IP Address of the SNTP Server: <input type="checkbox"/></p> <p>IP Address: <input type="text" value="0.0.0.0"/></p> <p>Subnet Mask: <input type="text" value="0.0.0.0"/></p> <p>MAC Address : 00:1a:e8:13:74:91</p> <p>Ethernet Link Mode: <input type="text" value="Auto"/></p> <p>Max. Data Packet Size (bytes): <input type="text" value="1500"/></p>

Expert mode >> Telephony Server >> Network Interface >> DHCP

Network Interfaces	
<p>Network Interfaces</p> <ul style="list-style-type: none"> LAN WAN DMZ FTP-Server DHCP 	<p>DHCP Mode</p> <p style="text-align: center;">Edit DHCP Mode</p> <p>No DHCP: <input checked="" type="radio"/></p> <p>DHCP Server: <input type="radio"/></p> <p>DHCP Relay Agent: <input type="radio"/></p>

Expert mode >> Telephony Server >> IP Routing >> Static Routes >> Default Router

“IP Address of Default Router” = XO customer site Router’s (IAD) internal IP.

Routing	
<ul style="list-style-type: none"> Routing ▼ IP Routing <ul style="list-style-type: none"> ▶ Static Routes Default Router DNS Server ▶ Gateway DNS Functionality Address Resolution Protocol ▶ ICMP Request ▶ IP Mapping ▶ NAT ▶ PSTN ▶ LCR E.164 	<div style="border: 1px solid #ccc; padding: 5px;"> <h3 style="margin: 0;">Default Router</h3> <p style="text-align: right; margin: 0;">Edit Default Router</p> <hr/> <p style="text-align: right;">Default Routing via: LAN</p> <p style="text-align: right;">IP Address of Default Router: 205.158.163.89</p> </div>

Expert mode >> Telephony Server >> IP Routing >> Gateway DNS functionality

Routing	
<ul style="list-style-type: none"> Routing ▼ IP Routing <ul style="list-style-type: none"> ▶ Static Routes Default Router DNS Server ▶ Gateway DNS Functionality Address Resolution Protocol ▶ ICMP Request ▶ IP Mapping ▼ NAT <ul style="list-style-type: none"> eMule eMule FTP-Server HTTP-Server MS Remote Desktop ▶ PSTN ▶ LCR E.164 	<div style="border: 1px solid #ccc; padding: 5px;"> <h3 style="margin: 0;">NAT</h3> <p style="text-align: right; margin: 0;"> Add NAT NAT Table Editor </p> <hr/> <p style="text-align: right;">NAT Rule Active: <input type="checkbox"/></p> <p style="text-align: right;">Description: <input style="width: 100%;" type="text"/></p> <p style="text-align: right;">Local IP Address: <input style="width: 100%;" type="text" value="0.0.0.0"/></p> <p style="text-align: right;">Local Port: <input style="width: 100%;" type="text" value="0"/></p> <p style="text-align: right;">Global Port: <input style="width: 100%;" type="text" value="0"/></p> <p style="text-align: right;">Protocol: UDP</p> </div>

Expert mode >> Telephony Server >> Voice Gateway >> SIP Parameters

Voice Gateway	
<ul style="list-style-type: none"> Voice Gateway H.323 Parameters SIP Parameters Codec Parameters ▶ Internet Telephony Service Provider ▶ Gatekeeper ▶ Destination Codec Parameters ▶ PBX ISDN Classmark 	<div style="border: 1px solid #ccc; padding: 5px;"> <h3 style="margin: 0;">SIP Parameters</h3> <p style="text-align: right; margin: 0;"> Edit SIP Parameters Edit SIP Registrar </p> <hr/> <p style="text-align: right;">SIP Transport Protocol</p> <p style="text-align: right;">SIP via TCP: Yes</p> <p style="text-align: right;">SIP via UDP: <input checked="" type="checkbox"/></p> <hr/> <p style="text-align: right;">Provider Calls</p> <p style="text-align: right;">Maximum possible Provider Calls: 7</p> </div>

Expert mode >> Voice Gateway >> XO-new

The screenshot shows the 'Voice Gateway' configuration page for 'XO-new'. The left sidebar lists various providers, with 'XO-new' selected. The main area is titled 'Internet Telephony Service Provider' and contains several sections:

- Internet Telephony Service Provider:** Includes fields for 'Provider Name' (XO-new), 'Enable Provider' (checked), 'Provider Identifier in System' (Provider 1), and 'Domain Name' (205.158.163.138). A red box highlights the 'Domain Name' field, with a red arrow pointing to it from the text 'NBS IP'.
- Provider Registrar:** Includes 'Use Registrar' (unchecked), 'IP Address / Host name' (0.0.0.0), 'Port' (0), and 'Reregistration Interval at Provider (sec)' (120).
- Provider Proxy:** Includes 'IP Address / Host name' (205.158.163.138) and 'Port' (5060).
- Provider Outbound Proxy:** Includes 'Use Outbound Proxy' (unchecked) and 'IP Address / Host name' (0.0.0.0).

At the bottom of the configuration area, there are buttons for 'Apply', 'Undo', 'Refresh', and 'Help'.

Domain Name: 205.158.163.138 (**XO - NBS IP**)

Provider proxy - IP Address/Host name: 205.158.163.138 (**XO - NBS IP**)

- PBX tested with SIP Trunk static registration configuration so “Registrar” settings are disabled.

SIP Provider profiles:

- A new SIP Provider profile was created for XO based on XO's requirements. Here are screen shots of that profile.

Application Diagnostics

- Application Diagnostics
 - Central Box
 - Developer-Settings
 - Trace Console Output
 - Take Over Write Token
 - SIP Provider Profiles
 - Mainboard
 - Slot 2

Wizard SIP Profile Configuration

Edit SIP Provider Profile

Provider Name: XO-new
 Serial number: 9876
 Domain Name: 205.158.163.138

Provider Registrar

IP Address / Host name: 0.0.0.0
 Port: 0
 Reregistration Interval at Provider (sec): 120
 Use Registrar:

Provider Proxy

IP Address / Host name: 205.158.163.138
 Port: 5060

Provider Outbound Proxy

Use Outbound Proxy:

Back OK & Next Delete Data Help

SIP Provider profiles (continued):

Application Diagnostics

Application Diagnostics

- ▼ Central Box
- ▼ Developer-Settings
 - Trace Console Output
 - Take Over Write Token
 - SIP Provider Profiles
- ▶ Mainboard
- ▶ Slot 2

Wizard SIP Profile Configuration

Domain Name:

Provider Registrar

IP Address / Host name:

Port:

Reregistration Interval at Provider (sec):

Use Registrar:

Provider Proxy

IP Address / Host name:

Port:

Provider Outbound Proxy

Use Outbound Proxy:

IP Address / Host name:

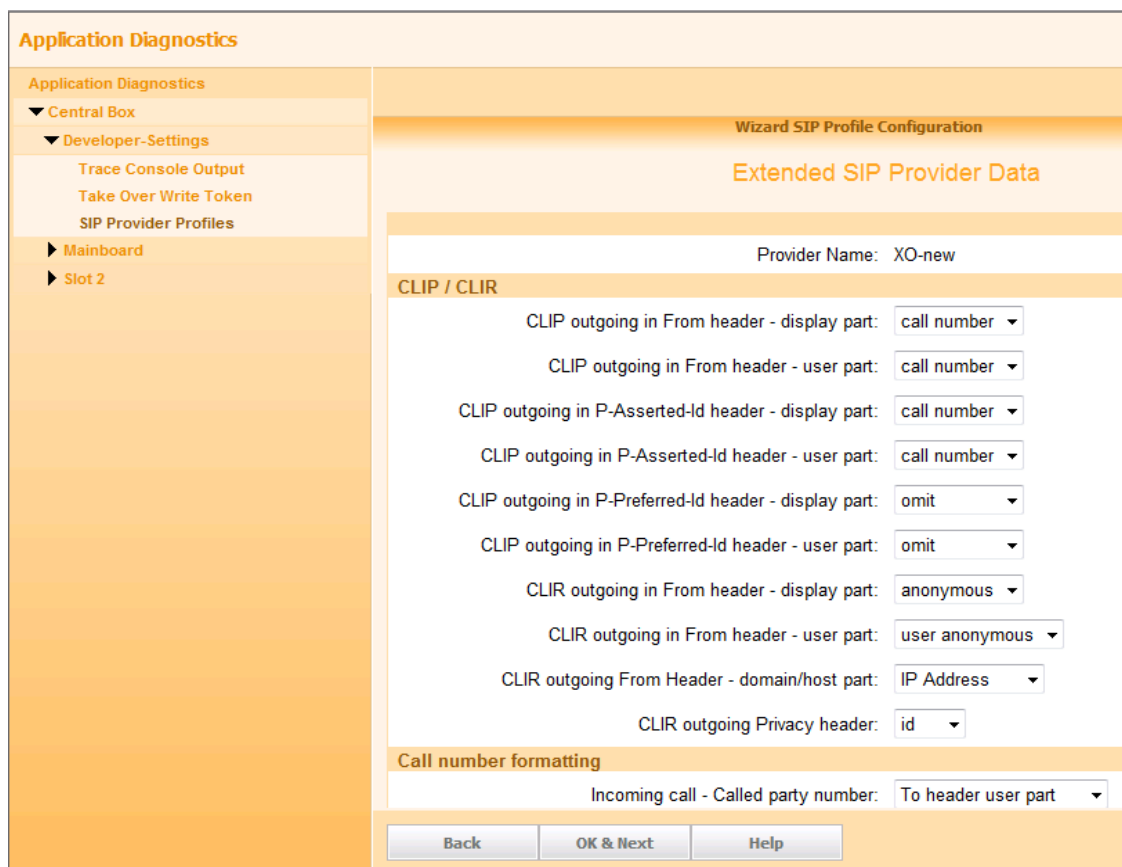
Port:

Provider STUN

Provider STUN IP Address:

Provider STUN Port Number:

SIP Provider profiles (continued):



Application Diagnostics

Application Diagnostics

- Central Box
- Developer-Settings
 - Trace Console Output
 - Take Over Write Token
 - SIP Provider Profiles
- Mainboard
- Slot 2

Wizard SIP Profile Configuration

Extended SIP Provider Data

Provider Name: XO-new

CLIP / CLIR

CLIP outgoing in From header - display part: call number

CLIP outgoing in From header - user part: call number

CLIP outgoing in P-Asserted-Id header - display part: call number

CLIP outgoing in P-Asserted-Id header - user part: call number

CLIP outgoing in P-Preferred-Id header - display part: omit

CLIP outgoing in P-Preferred-Id header - user part: omit

CLIR outgoing in From header - display part: anonymous

CLIR outgoing in From header - user part: user anonymous

CLIR outgoing From Header - domain/host part: IP Address

CLIR outgoing Privacy header: id

Call number formatting

Incoming call - Called party number: To header user part

Back OK & Next Help

- CLIP is configured as “call number” to display assigned DID of an extension as caller number in outbound call.
- CLIR is configured as “anonymous” in case if CLIR mode is enabled from extension. With this configuration outbound call number and name displayed as ‘anonymous’.
- Also with CLIR configuration, all “P-Asserted-id” header fields **must** be “call number” as shown in above screen shot.

SIP Provider profiles (continued):

- “RedirNrInFrom” flag is enabled to send out original caller number in “call forward” call scenario. Also this flag makes IPPBx to add “Diversion” header in such call.

Expert mode >> Telephony Server >> Voice Gateway >> Codec Parameters

For PKG1 (G711)

Voice Gateway

Voice Gateway
 H.323 Parameters
 SIP Parameters
 Codec Parameters
 ▶ Internet Telephony Service Provider
 ▶ Gatekeeper
 ▶ Destination Codec Parameters
 ▶ PBX
 ISDN Classmark

Codec Parameters

Edit Codec Parameters

Codec	Priority	Voice Activity Detection	Frame Size
G.711 A-law	Priority 7	VAD: <input type="checkbox"/>	30 msec
G.711 μ -law	Priority 1	VAD: <input type="checkbox"/>	20 msec
G.729A	not used	VAD: <input type="checkbox"/>	20 msec
G.729AB	not used	VAD: <input checked="" type="checkbox"/>	20 msec

T.38 Fax

T.38 Fax:
 Use FillBitRemoval:
 Max. UDP Datagram Size for T.38 Fax (bytes): 1472
 Error Correction Used for T.38 Fax (UDP): t38UDPRedundancy

Misc.

ClearChannel: Frame Size: 20 msec
 Transmission of Fax/Modem Tones according to RFC2833:
 Transmission of DTMF Tones according to RFC2833:
 Redundant Transmission of RFC2833 Tones according to RFC2198:

Expert mode >> Telephony Server >> Voice Gateway >> Codec Parameters

For PKG2 (G729, G711)

Voice Gateway

Voice Gateway
 H.323 Parameters
 SIP Parameters
 Codec Parameters
 ▶ Internet Telephony Service Provider
 ▶ Gatekeeper
 ▶ Destination Codec Parameters
 ▶ PBX
 ISDN Classmark

Codec Parameters

Edit Codec Parameters

Codec	Priority	Voice Activity Detection	Frame Size
G.711 A-law	Priority 7	VAD: <input type="checkbox"/>	30 msec
G.711 μ -law	Priority 2	VAD: <input type="checkbox"/>	20 msec
G.729A	Priority 1	VAD: <input type="checkbox"/>	20 msec
G.729AB	not used	VAD: <input checked="" type="checkbox"/>	20 msec

T.38 Fax

T.38 Fax:
 Use FillBitRemoval:
 Max. UDP Datagram Size for T.38 Fax (bytes): 1472
 Error Correction Used for T.38 Fax (UDP): t38UDPRedundancy

Misc.

ClearChannel: Frame Size: 20 msec
 Transmission of Fax/Modem Tones according to RFC2833:
 Transmission of DTMF Tones according to RFC2833:
 Redundant Transmission of RFC2833 Tones according to RFC2198:

Apply Undo Help

Expert mode >> Voice Gateway >> MSN (main number)

The screenshot displays the 'Voice Gateway' configuration window. On the left is a tree view of various providers, with 'XO-new' expanded to show a list of MSNs: 4693873321, 4693873322, 4693873323, and 4693873324. The main area is titled 'MSN Entry' and contains an 'Edit MSN Entry' tab and a 'Delete MSN Entry' button. Below this, the 'Internet Telephony Phone Number' section includes a text input field with '4693873321', a dropdown menu for 'Internal Call Number' set to '100 STATION 100', and a checked 'Default Entry' checkbox. At the bottom of the window are 'Apply', 'Undo', and 'Help' buttons.

Basic Settings:

Expert mode >> Telephony Server >> Basic Settings >> System Flags

Basic Settings

<div style="background-color: #f4a460; padding: 2px;">Basic Settings</div> <div style="background-color: #f4a460; padding: 2px;">▼ System</div> <div style="background-color: #f4a460; padding: 2px;">System Flags</div> <div style="background-color: #f4a460; padding: 2px;">Time Parameters</div> <div style="background-color: #f4a460; padding: 2px;">Display</div> <div style="background-color: #f4a460; padding: 2px;">DISA</div> <div style="background-color: #f4a460; padding: 2px;">Intercept/Attendant/Hotline</div> <div style="background-color: #f4a460; padding: 2px;">LDAP</div> <div style="background-color: #f4a460; padding: 2px;">Texts</div> <div style="background-color: #f4a460; padding: 2px;">Flexible menu</div> <div style="background-color: #f4a460; padding: 2px;">Speed Dials</div> <div style="background-color: #f4a460; padding: 2px;">Gateway</div> <div style="background-color: #f4a460; padding: 2px;">▶ DynDNS</div> <div style="background-color: #f4a460; padding: 2px;">AF/EF Codepoints</div> <div style="background-color: #f4a460; padding: 2px;">Quality of Service</div> <div style="background-color: #f4a460; padding: 2px;">▶ Date and Time</div> <div style="background-color: #f4a460; padding: 2px;">Port Management</div> <div style="background-color: #f4a460; padding: 2px;">▶ Call Charges</div> <div style="background-color: #f4a460; padding: 2px;">Access Points</div> <div style="background-color: #f4a460; padding: 2px;">Voicemail</div>	<div style="background-color: #f4a460; padding: 2px;">System Flags</div> <div style="background-color: #f4a460; padding: 2px; text-align: right;">Edit System Flags</div> <div style="padding: 10px;"> <p>Through-connection for external FWD on: <input checked="" type="checkbox"/></p> <p>Call forwarding to main station interface permitted: <input checked="" type="checkbox"/></p> <p>Hunting to external call forwarding destination: <input checked="" type="checkbox"/></p> <p style="padding-left: 100px;">Conference tone: <input type="checkbox"/></p> <p>Warning signal for call pickup groups: <input checked="" type="checkbox"/></p> <p>Increase volume for optiPoint/OpenStage terminals: <input type="checkbox"/></p> <p style="padding-left: 100px;">Relocate allowed: <input type="checkbox"/></p> <p>More than 1 external conference member: <input checked="" type="checkbox"/></p> <p style="padding-left: 100px;">Trunk reservation, automatic: <input type="checkbox"/></p> <p style="padding-left: 100px;">No. redial with a/c code: <input type="checkbox"/></p> <p>Use only default number for MSN : <input type="checkbox"/></p> <p style="padding-left: 100px;">Path optimization: <input checked="" type="checkbox"/></p> <p style="padding-left: 100px;">DTMF automatic: <input checked="" type="checkbox"/></p> <p>Broadcast with connection: <input checked="" type="checkbox"/></p> <p style="padding-left: 100px;">Tone from CO: <input type="checkbox"/></p> </div> <div style="background-color: #f4a460; padding: 5px; text-align: center;"> <input type="button" value="Apply"/> <input type="button" value="Undo"/> <input type="button" value="Help"/> </div>
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Expert mode >> Telephony Server >> Basic Settings >> System Flags (continued)

Basic Settings	
<p>Basic Settings</p> <ul style="list-style-type: none"> ▼ System <ul style="list-style-type: none"> System Flags Time Parameters Display DISA Intercept/Attendant/Hotline LDAP Texts Flexible menu Speed Dials Gateway ► DynDNS AF/EF Codepoints Quality of Service ► Date and Time Port Management ► Call Charges Access Points Voicemail 	<p>System Flags</p> <p style="text-align: right;">Edit System Flags</p> <p>Tone from CO: <input type="checkbox"/></p> <p>Ringback protection: <input type="checkbox"/></p> <p>Euro-impedance: <input type="checkbox"/></p> <p>Different phonemail messages Day/Night: <input type="checkbox"/></p> <p>Display international / national code number: <input type="checkbox"/></p> <p>Line change for direct call: <input checked="" type="checkbox"/></p> <p>Automatic redial: <input type="checkbox"/></p> <p>Voice mail node number: <input type="checkbox"/></p> <p>Call Pickup after automatic recall: <input type="checkbox"/></p> <p>Configurable CLIP: <input type="checkbox"/></p> <p>Caller list at destination in case of Forward Line: <input type="checkbox"/></p> <p>Call forwarding after deflect call / single step transfer: <input checked="" type="checkbox"/></p> <p>Follow call management in case of deflect call / single step transfer: <input checked="" type="checkbox"/></p> <p>E.164 numbering scheme: <input type="checkbox"/></p> <p>Calling number in pick-up groups / ringing groups / CFN /RNA: <input checked="" type="checkbox"/></p>

Expert mode >> Telephony Server >> Basic Settings >> System Flags (continued)

Basic Settings	
<p>Basic Settings</p> <ul style="list-style-type: none"> ▼ System <ul style="list-style-type: none"> System Flags Time Parameters Display DISA Intercept/Attendant/Hotline LDAP Texts Flexible menu Speed Dials Gateway ► DynDNS AF/EF Codepoints Quality of Service ► Date and Time Port Management ► Call Charges Access Points Voicemail 	<p>System Flags</p> <p style="text-align: right;">Edit System Flags</p> <p>Call forwarding after deflect call / single step transfer: <input checked="" type="checkbox"/></p> <p>Follow call management in case of deflect call / single step transfer: <input checked="" type="checkbox"/></p> <p>E.164 numbering scheme: <input type="checkbox"/></p> <p>Calling number in pick-up groups / ringing groups / CFN /RNA: <input checked="" type="checkbox"/></p> <p>SIP Prov. to SIP Prov. transit: <input type="checkbox"/></p> <p>Open numbering scheme</p> <p style="text-align: right;">active: <input type="checkbox"/></p> <p style="text-align: right;">Node number: <input type="text"/></p> <p>CO features (transfer/conf./drop)</p> <p style="text-align: right;">CO features: <input type="checkbox"/></p> <p>Transit permission</p> <p style="text-align: right;">Feature transit: <input checked="" type="checkbox"/></p> <p style="text-align: right;">Tie traffic transit: <input checked="" type="checkbox"/></p> <p style="text-align: right;">External traffic transit : <input checked="" type="checkbox"/></p> <p style="text-align: center;"> <input type="button" value="Apply"/> <input type="button" value="Undo"/> <input type="button" value="Help"/> </p>

Expert mode >> Telephony Server >> Basic Settings >> Display

Basic Settings

<div style="background-color: #f4a460; padding: 2px;">Basic Settings</div> <ul style="list-style-type: none"> ▼ System <ul style="list-style-type: none"> System Flags Time Parameters Display DISA Intercept/Attendant/Hotline LDAP Texts Flexible menu Speed Dials Gateway ▶ DynDNS AF/EF Codepoints Quality of Service ▶ Date and Time Port Management ▶ Call Charges Access Points Voicemail 	<div style="background-color: #f4a460; padding: 2px;">Gateway</div> <div style="background-color: #f4a460; padding: 2px; text-align: right; font-size: small;">Edit Gateway Properties</div> <div style="background-color: #f4a460; padding: 2px;">General</div> <table style="width: 100%; border-collapse: collapse;"> <tr><td style="width: 50%;">Customer name:</td><td><input type="text" value="MRB"/></td></tr> <tr><td>Contract number:</td><td><input type="text"/></td></tr> <tr><td>System Name:</td><td><input type="text" value="HiPath"/></td></tr> <tr><td>Gateway Location:</td><td><input type="text" value="SYR"/></td></tr> <tr><td>Contact Address:</td><td><input type="text"/></td></tr> <tr><td>System Country Code:</td><td><input type="text" value="USA"/> ▼</td></tr> <tr><td>Gateway IP Address:</td><td><input type="text" value="192.168.1.2"/></td></tr> <tr><td>Gateway Subnet Mask:</td><td><input type="text" value="255.255.255.0"/></td></tr> <tr><td>International Prefix:</td><td><input type="text" value="011"/></td></tr> <tr><td>National Prefix:</td><td><input type="text" value="1"/></td></tr> <tr><td>Brand:</td><td><input type="text" value="HiPath OpenOffice ME"/> ▼</td></tr> </table> <div style="background-color: #f4a460; padding: 2px;">Additional Features</div> <table style="width: 100%; border-collapse: collapse;"> <tr><td style="width: 50%;">Use RTP Proxy:</td><td><input checked="" type="checkbox"/></td></tr> </table>	Customer name:	<input type="text" value="MRB"/>	Contract number:	<input type="text"/>	System Name:	<input type="text" value="HiPath"/>	Gateway Location:	<input type="text" value="SYR"/>	Contact Address:	<input type="text"/>	System Country Code:	<input type="text" value="USA"/> ▼	Gateway IP Address:	<input type="text" value="192.168.1.2"/>	Gateway Subnet Mask:	<input type="text" value="255.255.255.0"/>	International Prefix:	<input type="text" value="011"/>	National Prefix:	<input type="text" value="1"/>	Brand:	<input type="text" value="HiPath OpenOffice ME"/> ▼	Use RTP Proxy:	<input checked="" type="checkbox"/>
Customer name:	<input type="text" value="MRB"/>																								
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Brand:	<input type="text" value="HiPath OpenOffice ME"/> ▼																								
Use RTP Proxy:	<input checked="" type="checkbox"/>																								

Expert mode >> Voice Gateway >> MSNs

The screenshot displays the 'Voice Gateway' configuration window. On the left is a tree view with the following structure:

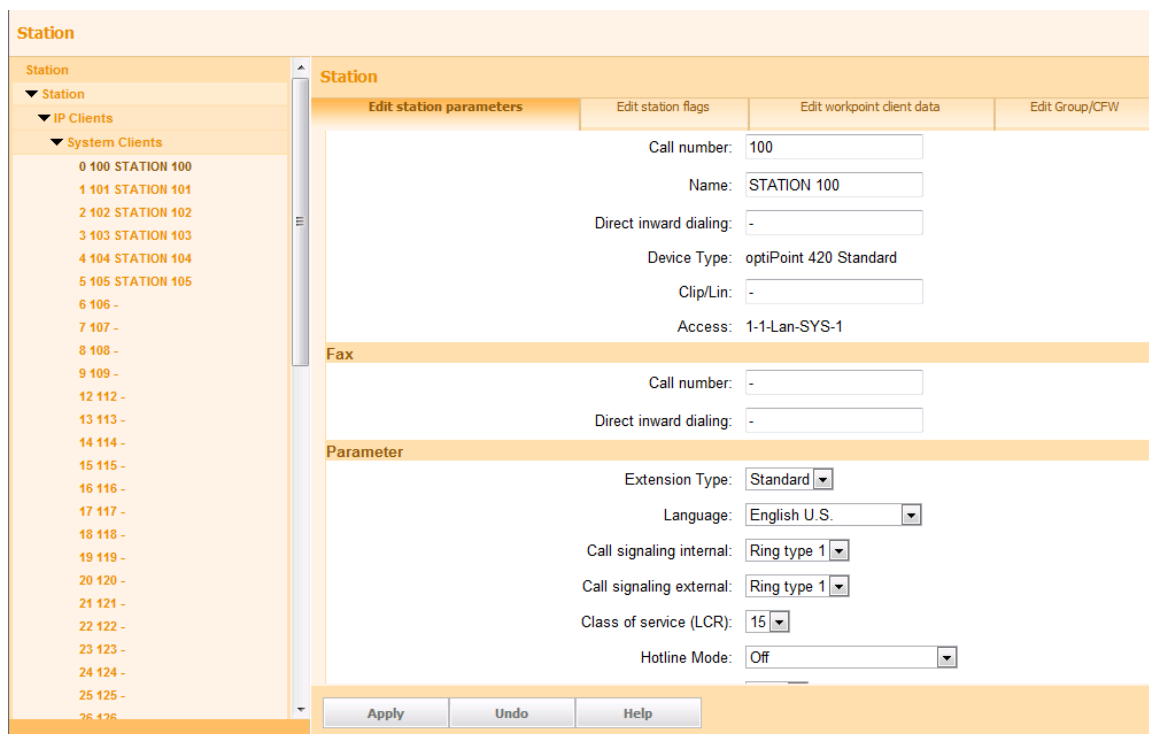
- ▶ PurTel
- ▶ QSC
- ▶ Sipgate
- ▶ SoTel with register
- ▶ SoTel
- ▶ STRATO
- ▶ T-Online
- ▶ TelePacific
- ▶ Telfree SA
- ▶ Telia
- ▶ Thinktel
- ▶ toplink
- ▶ Tus Telekom
- ▶ UPC Telemach
- ▶ VoiceFlex
- ▶ Vozelia
- ▶ Welcome Italia
- ▼ XO-new
 - ▼ 4693873320
 - ▼ MSNs
 - 4693873321
 - 4693873322
 - 4693873323
 - 4693873324
 - 4693873325
 - 4693873328
 - ▼ DID ranges
 - ▶ XSip

The main area is titled 'DID Range' and contains the following configuration options:

- Add DID Range** (button)
- Internet Telephony DID phone number range** (header)
 - PABX number:
 - Direct inward dialing band up :
 - to :
 - Begin of internal Call Number range:
 - Assign direct inward dialing band automatically to stations:

At the bottom of the window are three buttons: **Apply**, **Undo**, and **Help**.

Expert mode >> Station



Station

Station

- Station
 - IP Clients
 - System Clients
 - 0 100 STATION 100
 - 1 101 STATION 101
 - 2 102 STATION 102
 - 3 103 STATION 103
 - 4 104 STATION 104
 - 5 105 STATION 105
 - 6 106 -
 - 7 107 -
 - 8 108 -
 - 9 109 -
 - 12 112 -
 - 13 113 -
 - 14 114 -
 - 15 115 -
 - 16 116 -
 - 17 117 -
 - 18 118 -
 - 19 119 -
 - 20 120 -
 - 21 121 -
 - 22 122 -
 - 23 123 -
 - 24 124 -
 - 25 125 -
 - 26 126 -

Station

Edit station parameters	Edit station flags	Edit workpoint client data	Edit Group/CFW
Call number: 100			
Name: STATION 100			
Direct inward dialing: -			
Device Type: optiPoint 420 Standard			
Clip/Lin: -			
Access: 1-1-Lan-SYS-1			
Fax			
Call number: -			
Direct inward dialing: -			
Parameter			
Extension Type: Standard			
Language: English U.S.			
Call signaling internal: Ring type 1			
Call signaling external: Ring type 1			
Class of service (LCR): 15			
Hotline Mode: Off			

Apply Undo Help

- For Fax machine as an extension “Extension Type” is selected as ‘Fax’. With that make sure that “Class of Service (LCR)” has same value as other extension.
- For modem machine as an extension “Extension Type” is selected as “modem”. With that make sure that “Class of Service (LCR)” has same value as other extension.

Expert mode >> Station (continued)

Station

Station

- Station
 - IP Clients
 - System Clients
 - 0 100 STATION 100
 - 1 101 STATION 101
 - 2 102 STATION 102
 - 3 103 STATION 103
 - 4 104 STATION 104
 - 5 105 STATION 105
 - 6 106 -
 - 7 107 -
 - 8 108 -
 - 9 109 -
 - 12 112 -
 - 13 113 -
 - 14 114 -
 - 15 115 -
 - 16 116 -
 - 17 117 -
 - 18 118 -
 - 19 119 -
 - 20 120 -
 - 21 121 -
 - 22 122 -
 - 23 123 -
 - 24 124 -
 - 25 125 -
 - 26 126 -

Station

Edit station parameters	Edit station flags	Edit workpoint client data
	Override class of service on: <input type="checkbox"/>	
	Override Do Not Disturb: <input type="checkbox"/>	
	FWD external permitted: <input checked="" type="checkbox"/>	
	Prevention of voice calling off: <input checked="" type="checkbox"/>	
	Disa Class of service: <input type="checkbox"/>	
	Transit allowed via Hook-on: <input checked="" type="checkbox"/>	
	System telephone lock reset: <input type="checkbox"/>	
	MCID access: <input type="checkbox"/>	
	Entry in telephone directory: <input checked="" type="checkbox"/>	
	Edit tel. number: <input type="checkbox"/>	
	No group ringing on busy: <input type="checkbox"/>	
	Silent Monitoring: <input type="checkbox"/>	
	Associated dialing/services: <input type="checkbox"/>	
	Call waiting rejection on: <input checked="" type="checkbox"/>	
	Discreet Call: <input type="checkbox"/>	
	Discreet Call Lock: <input type="checkbox"/>	

Apply Undo Help

Hunt Group configuration:

Expert mode >> Incoming calls >> Groups/Hunt Groups

Incoming Calls

Incoming calls

- Groups/Hunt groups
 - (1)360 XO hunt
 - (1)100 STATION 100
 - (2)101 STATION 101
 - Mobility Entry groups
 - Team/top
 - Call pickup
 - UCD
 - Call Forwarding

Groups/Hunt groups

Edit Group Parameters	Delete Group	Display Members	Add Member
Group: 1			
Call number: 360			
Direct inward dialing number:			
Name: XO hunt			
Type: Linear hunt group			
Ring type: 1			
Tel. directory: <input type="checkbox"/>			

Apply Undo Help

Expert mode >> Incoming calls >> Groups/Hunt Groups >> Display Members

Incoming Calls

Incoming calls

- Groups/Hunt groups
 - (1)360 XO hunt
 - (1)100 STATION 100
 - (2)101 STATION 101
 - Mobility Entry groups
 - Team/top
 - Call pickup
 - UCD
 - Call Forwarding

Groups/Hunt groups

Edit Group Parameters Delete Group **Display Members** Add Member Edit Member Order

Group	Member	Call number	Name	Parameters	MULAP key set up	Route	Type
1	1	100	STATION 100		No		Linear hunt group
1	2	101	STATION 101		No		Linear hunt group

Expert mode >> Incoming calls >> Call Forwarding

Incoming Calls

Incoming calls

- Groups/Hunt groups
 - (1)360 XO hunt
 - (1)100 STATION 100
 - (2)101 STATION 101
 - Mobility Entry groups
 - Team/top
 - Call pickup
 - UCD
 - Call Forwarding

Call Forwarding

Edit Call Forwarding

Select Station / Group

100	STATION 100	
100	STATION 100	
101	STATION 101	
102	STATION 102	
103	STATION 103	

All stations
 Licensed stations
 All groups
 Arranged groups

Edit Call Forwarding

Call forwarding sequence

by day by night

first: 100 STATION 100

afterwards: User defined

afterwards: Voicemail box

afterwards:

The targets for day and night are always the same:

Destinations

Station

100	STATION 100
101	STATION 101
102	STATION 102
103	STATION 103

Groups

360	XO hunt
-	-
-	-

Announcements

Voicemail box

External destination

User defined

IP Phone OptiPoint420 Standard configuration:

SIEMENS
optiPoint420Standard IP address : 192.168.1.139
Tel : 100

[Home Page](#) [Administration User](#)

General Information	
MAC address:	00-1a-e8-01-43-09
Application version:	V5 R5.10.0
Web content version:	1.0
Phone identity:	34
Asset identity:	7-38-2-0-3-7183
Acoustic filter:	1
NetBoot version:	
Application Generic Platform version:	2.6

SIEMENS
optiPoint420Standard IP address : 192.168.1.139
Tel : 100

[Home Page](#) [Administration User](#)

WARNING:
Changing these settings may require you to update your browser's URL to access the phone.

Network and Routing	
DHCP:	<input type="checkbox"/>
Terminal IP address:	<input type="text" value="192.168.1.139"/>
Terminal mask:	<input type="text" value="255.255.255.0"/>
DNS addresses:	<input type="text" value="0.0.0.0"/> <input type="text" value="0.0.0.0"/> <input type="text" value="0.0.0.0"/>
Terminal hostname:	<input type="text" value="op420Std001ae8014309"/>
Domain Name:	<input type="text"/>
VLAN Method:	Manual ▾
Default VLAN ID:	<input type="text" value="0"/>

SIEMENS
optiPoint420Standard

IP address : 192.168.1.139
Tel : 100

[Home Page Administration User](#)

VLAN Method:
 Default VLAN ID:

IP Routing:
 Default Route:

Route 1: <input type="text" value="0.0.0.0"/>	Route 2: <input type="text" value="0.0.0.0"/>
Gateway 1: <input type="text" value="0.0.0.0"/>	Gateway 2: <input type="text" value="0.0.0.0"/>
Mask 1: <input type="text" value="0.0.0.0"/>	Mask 2: <input type="text" value="0.0.0.0"/>

Lan Port Settings:
 Lan Port 1 Mode:
 Lan Port 2 Mode:
 Lan Port 2 Enabled:
 Lan Port 2 Auto MDIX Enabled:

IP Phone OptiPoint420 Standard configuration for codec:

For PKG 1 (G.711)

optiPoint420Standard Home Page - Windows Internet Explorer

https://192.168.1.139/index.html

SIEMENS
optiPoint420Standard

IP address : 192.168.1.139
Tel : 100

[Home Page Administration User](#)

NOTE:
 All compression codecs will be offered, the 'Preferred compression' field sets which is offered first.
 The 'Packet size' field is not applicable to G.723, as this always uses 30 ms.
 The 'Min. Jitter Buffer Target' is recommended to be 30ms unless minimising latency is a priority.

Audio Settings	
Home	Standby
Codec: <input type="text" value="High quality preferred"/> <input type="button" value="v"/>	Codec: <input type="text" value="High quality preferred"/> <input type="button" value="v"/>
Preferred compression: <input type="text" value="G.723"/> <input type="button" value="v"/>	Preferred compression: <input type="text" value="G.729"/> <input type="button" value="v"/>
Silence suppression: <input type="checkbox"/>	Silence suppression: <input type="checkbox"/>
Packet size: <input type="text" value="Auto"/> <input type="button" value="v"/>	Packet size: <input type="text" value="Auto"/> <input type="button" value="v"/>
G722 codec: <input type="checkbox"/>	G722 codec: <input type="checkbox"/>
Min. Jitter Buffer Target: <input type="text" value="30 ms"/> <input type="button" value="v"/>	Min. Jitter Buffer Target: <input type="text" value="30 ms"/> <input type="button" value="v"/>

For PKG 2 (G.729, G.711)

SIEMENS
optiPoint420Standard

IP address : 192.168.1.139
Tel : 100

[Home Page Administration User](#)

NOTE:

All compression codecs will be offered, the 'Preferred compression' field sets which is offered first.
The 'Packet size' field is not applicable to G.723, as this always uses 30 ms.
The 'Min. Jitter Buffer Target' is recommended to be 30ms unless minimising latency is a priority.

Audio Settings

Home	Standby
Codec: Low bandwidth preferred ▼	Codec: Low bandwidth preferred ▼
Preferred compression: G.729 ▼	Preferred compression: G.729 ▼
Silence suppression: <input type="checkbox"/>	Silence suppression: <input type="checkbox"/>
Packet size: Auto ▼	Packet size: Auto ▼
G722 codec: <input type="checkbox"/>	
Min. Jitter Buffer Target: 30 ms ▼	
<input type="button" value="Submit"/> <input type="button" value="Reset"/>	