SIP Verification Test Plan and Report for *GrandStream GXP-2000 & Budgetone 100*

Test Result	PASS
Test Date	
Product Name	GXP-2000; Budgetone 100
Product Version # (must be generally available)	
CallManager Version X.X(x)	5.0(X)
Cisco Security Agent Version (CSA)	
Product Type(Billing, Voice Recording, phone apps etc):	
API/Protocol(s) Used	
Developer Services Contract #	
Partner IVT Contact Name:	
Partner IVT Contact Phone:	
Partner IVT Contact Email:	
IVT Lab Location (EMEA or US):	
Partner Main Support Number	
Partner Main Support Email	

Reviewers

Department	Name
Product Marketing Manager	Larry Schessel
Product Manager	Steve Levy
Software Development Manager	Mark Atkinson
PBX Interoperability Lab	Patty Mertz
Marketing Program Manager	James Stormes
Product Marketing Manager	John Lamarque
Product Manager	David Kelly

Modification History

Rev	Date	Originator	Comment
1	1/25/2006	Samir Batio	Initial Document Draft
2	2/14/2006	Samir Batio	Updated Test Results
3	2/17/2006	Samir Batio	Updated Test Results for Granstream BT100 and GXP-2000 phones.

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1 Introduction

This test plan template will be used to record test results for 3rd party vendor SIP Phones' Interoperability testing with Cisco CallManager Seadragon-GA release. Due to the lack of Multi-Vendor SIP Interoperability Testing standards, it is necessary to verify some level of interoperability between endpoint devices and SIP features supported in CallManager 5.0 Seadragon-GA. The purpose of this testing is to verify that Cisco CallManager 5.0 can support 3rd-party SIP phones, and to also ascertain what features are/are not supported.

1.1 Scope

The scope of this test plan is:

- Test the Cisco CCM 5.0 Seadragon-GA release Interoperability to other vendors' SIP Phones for Basic and Supplementary services and list any feature limitations.

Features Test Coverage:

- SIP Phone Registration
- Basic Calls between Cisco SIP phones and other vendors' SIP Phones.
- Basic Calls between Cisco IP phones (SCCP) on CCM side and other vendors' SIP phones.
- Basic Calls with Calling/Connected Name and Number Identification Presentation.
- Alerting Name
- Hold and Resume.
- Call Transfer (Blind, Attended, Early Attended)
- Call Forwarding (CFA-Call Forwarding All, CFB-Call Forwarding on Busy, CFNA-Call Forwarding No Answer)
- Call Conferencing-Basic 3-way call.
- MWI Support

List of 3rd Party phones that will be tested:

- Grandstream Budgetone 100
- Grandstream GXP-2000

List of Cisco phones that will be tested:

- Cisco 7960 SCCP
- Cisco 7961 SIP

As much as possible, functional tests that could be tested at the user interface are described in this document. The deployment of a protocol monitor/simulator is needed to observe the actual signaling information when tests cannot be successfully completed. Each feature is described in a separate section with a brief description, test configuration and administration required before the actual test cases are described.

1.2 Background for Compatibility/Interoperability Tests

For these test cases, there will be an MCS-78XX server configured with CM 5.0. There will be Cisco 7960 SCCP-controlled phones, as well as Cisco 7961 SIP phones, registered with the CallMananger server. 3rd-party vendor SIP phones will be configured to attempt registration with CallManager. After validating successful SIP registration, basic calls will be placed from/to each phone, to validate its functionality/compatibility/interoperability.

1.3 References

- RFC2833, RFC3261, RFC3262, RFC3263, RFC3264, RFC3265
- RFC 3311, RFC 3312, RFC 3326, RFC 3420, RFC 3515

1.4 Terminology

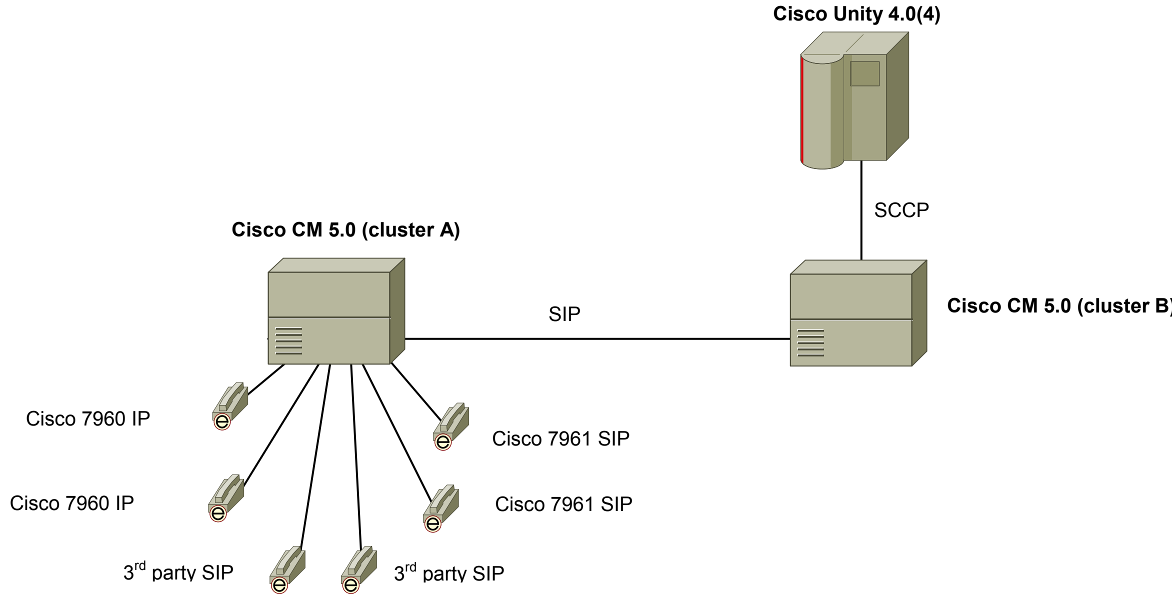
1.4.1 General

CCM	Cisco CallManager
SIP	Session Initiation Protocol
Supplementary Service	Services beyond voice or data connectivity, number transport and display. Examples include call forwarding, transfer, and call hold.
Voice path	The two-way communication path between parties on a call. For station to station calls, voice path is verified by simply ensuring that what is spoken into Station A can be heard at Station B, and vice versa.
BT100	Budgetone 100 Series phone from Grandstream

2 Summary Test report

2.1 Network Diagram

The network diagram below illustrated is a simplified view of the test bed used for all the test cases mentioned in this document:



2.2 Test result matrix

The table below is a summary of results of CallManager 5.0 interoperability testing with other vendor's SIP phones for all available SIP features.

Features Tested	Vendor SIP phone Test Results		Summary
	Grandstream BT 100	**Grandstream GXP-2000**	
SIP Phone Registration	Pass	Pass	
SIP Phone Registration-Multiple Lines	Not Available	Pass	
Basic Calls	Pass (read summary	Pass	On Grandstream BT100 phone, speaker is not turned off upon remote disconnect and a special cadence tone is played indicating that the far end disconnected the call.
Calling Name Identification	Pass	Pass	Grandstream Phones do not support sending Calling Name using the Remote Party ID header but rather they send it in the From header. Cisco SIP Phone however will not display the Calling Name of the Grandstream phone but will display the Caller ID information for the Grandsteam Phone that is configured in CCM's DN for that particular phone.
Connected Name Identification	Not Supported	Not Supported	CCM sends Connected Name information using the Remote Party ID header. Grandstream (BT100 as well as GXP-2000) SIP phones did not display Connected Name.
Alerting Name Identification	Not Supported	Not Supported	CCM sends the Alerting Name information using the Remote Party ID header in the 180 Ringing SIP response message but Grandstream (both Budgetone 100 and GXP-2000) phones did not display the Alerting Name on their display.

Hold and Resume	Pass	Pass	For 3rd Party SIP phone initiated calls, when the call is put on hold either from 3rd party phone side or Cisco side, the last digit of the calling number displayed on the Cisco phone is missing. This is ccm issue CSCsc96611
Attended Call Transfer	Pass (read summary)	Pass (read summary)	In order to support Attended Call Transfer on the Grandstream phones (both BT100 and GXP-2000), the MTP Required check box needs to be checked in the corresponding CCM's Phone configuration.
Early Attended Call Transfer	Not Supported	Not Supported	Early Attended Call Transfer is not supported by Grandstream phones. Only Attended and Blind Call Transfers are supported
Blind Call Transfer	Pass	Pass	
Call Forwarding All	Pass	Pass	
Call Forwarding No Answer	Pass	Pass	
Call Forwarding Busy	Pass	Pass	To trigger CFB from the GXP-2000 phone instead of CCM, we have to set the busy trigger for the CCM's DN for that particular phone higher than 11 since the GXP-2000 phone can support up to 11 calls on same line.
Call Conference	Pass	Pass	
Voicemail Access and MWI support	Pass	Pass	
DTMF tone outpulsing to Cisco Unity ports	Pass	Pass	

2.3 Test result Summary

These test results are valid for CCM5.0 Seadragon release only.

- 3rd Party SIP Phones cannot be reset from Cisco CallManager. Every time they need a reset, it has to be done manually from the phone.
- 3rd Party SIP Phones cannot Auto Register with Cisco CallManager. They need to be manually added.
- CallManager will challenge the 3rd Party phone upon attempt to register for user ID and password and if the phone does not respond with the proper credentials then CCM will reject the registration.
- If digest authentication is disabled on SIP phone security profile, CCM will challenge the phone for user ID only.
- If digest authentication is Enabled on SIP phone security profile, CCM will challenge the phone for user ID and Password, phone password must match the Digest credentials entered on the end user page.
- Every 3rd Party SIP Phone would require a separate Digest User to be created for it to be authenticated.
- 3rd Party SIP Phones cannot download their configuration from Call Manager TFTP server, so someone needs to configure them manually.
- For 3rd Party SIP phone initiated calls, when the call is put on hold either from 3rd party phone side or Cisco side, the last digit of the calling number displayed on the Cisco phone is missing. This is ccm issue CSCsc96611.
- In order to support Attended Call Transfer on the Grandstream (both BT100 and GXP-2000) phones, the “MTP Required” check box needs to be checked in the CCM’s Phone configuration for that particular phone. Furthermore the Attended transfer should be done from the same line number (for GXP-2000 with multi-line configuration) on the Grandstream GXP-2000 phone since CCM does not support transferring a call on one line to a call on another line. The calls have to be on the same line.
- To trigger CFB from the Grandstream GXP-2000 phone instead of CCM, we have to set the busy trigger on CCM’s DN for that particular phone higher than 11 since the GXP-2000 phone can support up to 11 line appearances.

3 SIP Phone Registration

3.1 Test Cases for SIP Phone Registration

3.1.1 SIP phone Registration

Action: Consider the following table for all necessary administration parameters

Test Case Iteration	Power up phone	Configure it to register with CM	Results
Grandstream Budgetone 100 Series			Pass
Grandstream GXP-2000			Pass

Action: Add a new SIP Phone to CM.

Action: Power up SIP Phone. Configure necessary registration parameters on phone.

Verification(1): Verify that the SIP phone are registered on the CM and that the CCM Serviceability Real-time monitoring Tool Device counter associated with SIP Phones is incremented accordingly.

Result: *passed?*___X_____*failed?*_________

Comments:

3.1.2 SIP phone Unregistration

Test Case Iteration	Results
Grandstream Budgetone 100 Series	Pass
Grandstream GXP-2000	Pass

Action: Disconnect a registered SIP Phone.

Verification(1): Verify that the SIP Phone is shown as Unregistered using the CCM Serviceability Real-Time Monitoring Tool, and that proper device counter is decremented.

Action: Plug phone back in, wait for phone registration. After phone registration, delete phone in CallManager.

Verification(2): Verify that phone becomes unregistered and is no longer operational.

Result: *passed?*___X_____*failed?*_________

Comments:

3.1.3 SIP phone Registration – Multiple Lines

Test Case Iteration	Results
Grandstream Budgetone 100 Series	Not Applicable
Grandstream GXP-2000	Pass, see comments

Action: Add a new SIP Phone to CM, and configure 2 phone extensions on it.

Action: Power up SIP Phone. Configure necessary registration parameters on phone.

Verification(1): Verify that the SIP phone is registered on the CM and that the CCM Serviceability Device counter associated with SIP Phones is incremented accordingly. Make sure both lines are operational.

Result: *passed?___X____failed?_________*

Comments:
Grandstream GXP-2000 was configured with 2 lines. CCM phone cofig shows Partial Registered eventhough both lines were registered and operational.

3.1.4 SIP phone support of HTTP Digest

Test Case Iteration	Configure it to register with CM	Results
Grandstream Budgetone 100 Series		Pass
Grandstream GXP-2000		Pass

Action: Enable HTTP Digest on CallManager.

Action: Configure SIP Phone on CallManager.

Action: Power up SIP Phone. Configure necessary registration parameters on phone, including any parameter enabling HTTP Digest.

Verification(1): Verify that the SIP phone is registered on the CM and that the CCM Serviceability Device counter associated with SIP Phones is incremented accordingly

Result: *passed?___X_____failed?_________*

Comments:

4 Basic Call

4.1 Test Cases for Basic Call

4.1.1 Station-to-Station call SIP call

Test Case Iteration	Place call to Cisco SIP Phone	Place call from Cisco SIP Phone	Results
Grandstream Budgetone 100 Series	Pass	Pass	Pass, see comment
Grandstream GXP-2000	Pass	Pass	Pass

Action: Call from Vendor SIP Phone to Cisco SIP Phone and vice versa.

Action: Answer call.

Action: Disconnect call.

Verification(1): validate that talk path has been established between phones.

Verification(2): validate that both ends of connection are properly torn down, and phones go back to idle state.

Result: *passed?__X______failed?_________*

Comments:
For Grandstream BT100 phone:
Verification(2): When the Cisco SIP phone clears the call, the call is properly torn down and both Cisco phone as well as Grandstream BT100 phone go to idle state. However on Grandstream BT100 phone, speaker is not turned off upon remote disconnect and a special cadence tone is played indicating that the far end disconnected the call.

4.1.2 Incoming and Outgoing Voice Calls with Line Identification

These test cases test passing of line identification display.

4.1.2.1 Voice call from vendor SIP phone to Cisco SIP Phone – number presentation

Test Case Iteration	Send Calling Number	Send Connected Number	Results
Grandstream Budgetone 100 Series	Yes	Yes	Pass, see comments
Grandstream GXP-2000	Yes	Yes	Pass, see comments

Action: Make a voice call from vendor SIP phone to Cisco SIP Phone

Verification(1): Talk path is available.

Verification(2): The calling number is correctly displayed on phones.

Verification(3): The connected number is displayed on phones

Action: Hang up and check the status of the call.

Verification(4): The call is cleared.

Result: *passed?*___*X*_____*failed?*_________

Comments:
Verification(3): Both Grandsteam SIP phones (Budgetone 100 and GXP-2000) do not display Connected Number. They display dialed number instead.

4.1.2.2 Voice call from Cisco SIP Phone to vendor SIP phone – number presentation

Test Case Iteration	Send Calling Number	Send Connected Number	Results
Grandstream Budgetone 100 Series	Yes	Yes	Pass
Grandstream GXP-2000	Yes	Yes	Pass

Action: Make a voice call from Cisco SIP Phone to vendor SIP phone

Verification(1): Talk path is available.

Verification(2): The calling number is correctly displayed on phones.

Verification(3): The connected number is displayed on phones

Action: Hang up and check the status of the call.

Verification(4): The call is cleared.

Result: *passed?*___X_____*failed?*_________

Comments:

4.1.2.3 Voice call from vendor SIP phone to Cisco IP Phone (SCCP) – number presentation

Test Case Iteration	Send Calling Number	Send Connected Number	Results
Grandstream Budgetone 100 Series	Yes	Yes	Pass, see comments
Grandstream GXP-2000	Yes	Yes	Pass, see comments

Action: Make a voice call from vendor SIP Phone to Cisco IP phone (SCCP).

Verification(1): Talk path is available.

Verification(2): The calling number is correctly displayed on phones.

Verification(3): The connected number is displayed on phones.

Action: Hang up and check the status of the call.

Verification(4): The call is cleared.

Result: *passed?*___X_____*failed?*_________

Comments:
Verification(3): Both Grandsteam SIP phones (Budgetone 100 and GXP-2000) do not display Connected Number. They display dialed number instead.

4.1.2.4 Voice call from Cisco IP Phone (SCCP) to vendor SIP phone– number presentation

Test Case Iteration	Send Calling Number	Send Connected Number	Results
Grandstream Budgetone 100 Series	Yes	Yes	Pass
Grandstream GXP-2000	Yes	Yes	Pass

Action: Make a voice call from Cisco IP phone (SCCP) to vendor SIP Phone.

Verification(1): Talk path is available.

Verification(2): The calling number is correctly displayed on phones.

Verification(3): The connected number is displayed on phones.

Action: Hang up and check the status of the call.

Verification(4): The call is cleared.

Result: passed?____X____failed?_________

Comments:

4.1.3 Unsuccessful Call Termination and Call Clearing

These tests verify voice call procedures when clearing is initiated during the establishment phase. They Also verify the tone strategy for each manufacturer.

4.1.3.1 Outgoing Call vendor SIP to Cisco SIP - No Answer

Test Case Iteration	Results
Grandstream Budgetone 100 Series	Pass
Grandstream GXP-2000	Pass

Action: Make a call from vendor phone to Cisco SIP Phone, but do not answer call.

Verification(1): Verify that the caller hears ring back and that the destination terminal rings.

Action: calling party should then hang up.

Verification(2): Call is cleared when the calling party goes on hook.

Result: passed?___X_____failed?_________

Comments:

4.1.3.2 Outgoing Call Cisco SIP to vendor SIP - No Answer

Test Case Iteration	Results
Grandstream Budgetone 100 Series	Pass
Grandstream GXP-2000	Pass

Action: Make a call from Cisco SIP Phone to vendor phone, but do not answer call.

Verification(1): Verify that the caller hears ring back and that the destination terminal rings.

Action: calling party should then hang up.

Verification(2): Call is cleared when the calling party goes on hook.

Result: passed?___X_____failed?_________

Comments:

4.1.3.3 Outgoing Call vendor SIP to Cisco SIP - Busy Station

Test Case Iteration	Results
Grandstream Budgetone 100 Series	Pass
Grandstream GXP-2000	Pass

Action: Busy out Cisco SIP Phone.

Action: Make a voice call from vendor SIP Phone to the busy station.

Verification(1): Calling user receives busy indication.

Action: Hang-up.

Result: *passed?___X_____failed?________*

Comments:

4.1.3.4 Outgoing Call Cisco SIP to vendor SIP - Busy Station

Test Case Iteration	Results
Grandstream Budgetone 100 Series	Pass
Grandstream GXP-2000	Pass

Action: Busy out vendor SIP Phone.

Action: Make a voice call from Cisco SIP Phone to the busy station.

Verification(1): Calling user receives busy indication.

Action: Hang-up.

Result: *passed?___X_____failed?________*

Comments:

4.1.3.5 Outgoing Call vendor SIP to Cisco SCCP - No Answer

Test Case Iteration	Results
Grandstream Budgetone 100 Series	Pass
Grandstream GXP-2000	Pass

Action: Make a call from vendor SIP Phone to Cisco IP Phone (SCCP), but do not answer call.

Verification(1): Verify that the caller hears ring back and that the destination terminal rings.

Action: calling party should then hang up.

Verification(2): Call is cleared when the calling party goes on hook.

Result: *passed?___X_____failed?________*

Comments:

4.1.3.6 Outgoing Call Cisco SCCP to vendor SIP - No Answer

Test Case Iteration	Results
Grandstream Budgetone 100 Series	Pass
Grandstream GXP-2000	Pass

Action: Make a call from Cisco SCCP IP Phone to vendor SIP Phone, but do not answer call.

Verification(1): Verify that the caller hears ring back and that the destination terminal rings.

Action: calling party should then hang up.

Verification(2): Call is cleared when the calling party goes on hook.

Result: *passed?___X_____failed?_________*

Comments:

4.1.3.7 Outgoing Call vendor SIP to Cisco SCCP - Busy Station

Test Case Iteration	Results
Grandstream Budgetone 100 Series	Pass
Grandstream GXP-2000	Pass

Action: Make Cisco SCCP station busy.

Action: Make a voice call from vendor SIP phone to Cisco SCCP IP Phone.

Verification(1): Calling user receives busy indication.

Action: Hang-up.

Result: *passed?___X_____failed?_________*

Comments:

4.1.3.8 Outgoing Call Cisco SCCP to vendor SIP - Busy Station

Test Case Iteration	Results
Grandstream Budgetone 100 Series	Pass
Grandstream GXP-2000	Pass

Action: Make vendor SIP station busy.

Action: Make a voice call from Cisco SCCP IP Phone to vendor SIP phone.

Verification(1): Calling user receives busy indication.

Action: Hang-up.

Result: *passed?___X_____failed?_________*

Comments:

5 Calling/Connected Name Identification

5.1 Description

These tests should be done only if both manufacturers support Name Identification.

5.2 Test Cases

5.2.1 Vendor SIP to Cisco SIP Voice call with Calling/Connected Name Identification

Test Case Iteration	Calling Name	Connected Name
Grandstream Budgetone 100 Series	Yes, see comments	No
Grandstream GXP-2000	Yes, see comments	No

Action: Place a call from vendor SIP Phone to Cisco SIP Phone.

Action: Answer the call

Verification(1): Verify Talk path is available.

Verification(2): Verify the calling name display on receiving station is in accordance to phone's administration.

Verification(3): Verify the connected name display on originating station is in accordance to phone's administration.

Result: *passed?___X_____failed?________*

Comments:
Verification(2): Grandstream Phones do not support sending Calling Name in the Remote Party ID header but rather they send it in the From header. Cisco SIP Phone however will not display the Calling Name of the Grandstream phone but will display the Caller ID information for the Grandsteam Phone that is configured in CCM's DN for that particular phone.

Verification(3): Grandstream phones did not display Connected Name.

5.2.2 Cisco SIP to vendor SIP Voice call with Calling/Connected Name Identification

Test Case Iteration	Calling Name	Connected Name
Grandstream Budgetone 100 Series	No	Yes, see comments
Grandstream GXP-2000	Yes	Yes, see comments

Action: Place a call from Cisco SIP Phone to vendor SIP Phone.

Action: Answer the call

Verification(1): Verify Talk path is available.

Verification(2): Verify the calling name display on receiving station is in accordance to phone's administration.

Verification(3): Verify the connected name display on originating station is in accordance to phone's administration.

Result: *passed?___X_____failed?________*

Comments:
Verification(2): Grandstream Budgetone 100 SIP phone did not display Calling Name eventhough Cisco SIP phone did send the Calling Name.
Verification(3): Cisco SIP phone displayed the connected name of the Grandstream phone that is configured in CCM's DN for that particular phone.

5.2.3 Vendor SIP to Cisco SCCP Voice call with Calling/Connected Name Identification

Test Case Iteration	Calling Name	Connected Name
Grandstream Budgetone 100 Series	Yes, see comments	No
Grandstream GXP-2000	Yes	No

Action: Place a call from vendor SIP Phone to Cisco IP Phone (SCCP).

Action: Answer the call

Verification(1): Verify Talk path is available.

Verification(2): Verify the calling name display on receiving station is in accordance with phone's administration.

Verification(3): Verify the connected name display on originating station is in accordance with phone's administration.

Result: passed?___X_____failed?________

Comments:
Verification(2): Grandstream Phones do not support sending Calling Name in the Remote Party ID header but rather they send it in the From header. Cisco Phone however will not display the Calling Name of the Grandstream phone but will display the Caller ID information for the Grandsteam Phone that is configured in CCM's DN for that particular phone.

Verification(3): CCM sends Connected Name information in the Remote Party ID header. Grandstream (BT100 as well as GXP-2000) SIP phone did not display Connected Name.

5.2.4 Cisco SCCP to vendor SIP Voice call with Calling/Connected Name Identification

Test Case Iteration	Calling Name	Connected Name

Grandstream Budgetone 100 Series	Not Supported	Yes, see comments
Grandstream GXP-2000	Yes	Yes, see comments

Action: Place a call from Cisco SCCP IP Phone to vendor SIP Phone.

Action: Answer the call

Verification(1): Verify Talk path is available.

Verification(2): Verify the calling name display on receiving station is in accordance with phone's administration.

Verification(3): Verify the connected name display on originating station is in accordance with phone's administration.

Result: passed?____X____failed?________

Comments:
Verification(2): Grandstream Budgetone 100 SIP phone did not display Calling Name.
Verification(3): Cisco SIP phone displayed the connected name of the Grandstream phone that is configured in CCM's DN for that particular phone

6 Alerting Name Identification

6.1 Description

This optional service provides for indicating name information to called parties during call set-up, which can be different than the actual calling/connected name.

6.2 Test Cases

6.2.1 Vendor SIP to Cisco SIP Voice call with Alerting Name enabled

Test Case Iteration	Pass/Fail	Phone Display
Grandstream Budgetone 100 Series	Not Supported	No alerting name
Grandstream GXP-2000	Not Supported	No alerting name

Action: Enable/Configure Alerting Name on Cisco SIP Phone.

Action: Place call from vendor SIP Phone to Cisco SIP Phone.

Verification(1): Verify the alerting name displayed on vendor SIP Phone is in accordance with the administration as indicated above. Record phone display results in table above.

Result: passed?___X____failed?________

Comments:
Verification(1): CCM sends the Alerting Name in the Remote Party ID header in the 180 Ringing SIP response but Grandstream (both Budgetone 100 and GXP-2000) phones did not display the Alerting Name on their display.

6.2.2 Vendor SIP to Cisco SCCP Voice call with Alerting Name enabled

Test Case Iteration	Pass/Fail	Phone Display
Grandstream Budgetone 100 Series	Not Supported	No alerting name
Grandstream GXP-2000	Not Supported	No alerting name

Action: Enable/Configure Alerting Name on Cisco SCCP IP Phone.

Action: Place call from vendor SIP Phone to Cisco SCCP IP Phone.

Verification(1): Verify the alerting name displayed on vendor SIP Phone is in accordance with the administration as indicated above. Record phone display results in table above.

Result: passed?___X_____failed?________

Comments:
Verification(1): CCM sends the Alerting Name in the Remote Party ID header in the 180 Ringing SIP response but Grandstream (both Budgetone 100 and GXP-2000) phones did not display the Alerting Name on their display.

6.2.3 Cisco SCCP to vendor SIP Voice call with Alerting Name enabled

Test Case Iteration	Pass/Fail	Phone Display
Grandstream Budgetone 100 Series	Pass	"MERCURY-25(A)"
Grandstream GXP-2000	Pass	"MERCURY-26(A)"

Action: Configure Alerting Name on vendor SIP Phone.

Action: Place call from Cisco SCCP IP Phone to vendor SIP Phone.

Verification(1): Verify the alerting name displayed on Cisco SCCP IP Phone is in accordance with the administration as indicated above. Record phone display results in table above.

Result: passed?___X_____failed?_________

Comments:

6.2.4 Vendor SIP to Cisco SIP Voice call with Alerting Name restricted

Test Case Iteration	Pass/Fail	Phone Display
Grandstream Budgetone 100 Series	Not Applicable	No alerting name
Grandstream GXP-2000	Not Applicable	No alerting name

Action: Configure Alerting Name to be restricted on Cisco SIP Phone.

Action: Place call from vendor SIP Phone to Cisco SIP Phone.

Verification(1): Verify the alerting name is not displayed on vendor SIP Phone in accordance with the administration as indicated above. Record phone display results in table above.

Result: passed?___X_____failed?_________

Comments:
Verification(1): Grandstream (both Budgetone 100 and GXP-2000) phones do not support Alerting Name presentation on their display, therefore this test case is not applicable.

6.2.5 Cisco SIP to vendor SIP Voice call with Alerting Name restricted

Test Case Iteration	Pass/Fail	Phone Display
Grandstream Budgetone 100 Series	Not Supported	"MERCURY-25(A)"
Grandstream GXP-2000	Not Supported	"MERCURY-26(A)"

Action: Configure Alerting Name to be restricted on vendor SIP Phone.

Action: Place call from Cisco SIP Phone to vendor SIP Phone.

Verification(1): Verify the alerting name is not displayed on Cisco SIP Phone in accordance with the administration as indicated above. Record phone display results in table above.

Result: passed?___X_____failed?________

Comments: Alerting Name Restriction not supported on Grandstream SIP phones.

6.2.6 Cisco SCCP to vendor SIP Voice call with Alerting Name restricted

Test Case Iteration	Pass/Fail	Phone Display
Grandstream Budgetone 100 Series	Not Supported	"MERCURY-25(A)"
Grandstream GXP-2000	Not Supported	"MERCURY-26(A)"

Action: Configure Alerting Name to be restricted on vendor SIP Phone.

Action: Place call from Cisco SCCP IP Phone to vendor SIP Phone.

Verification(1): Verify the alerting name is not displayed on Cisco SCCP Phone in accordance with the administration as indicated above. Record phone display results in table above.

Result: passed?___X_____failed?_______

Comments: Alerting Name Restriction not supported on Grandstream SIP phones.

7 Hold & Resume

7.1 Test Cases for Hold & Resume

7.1.1 Vendor SIP to Cisco SIP Voice call – originating end initiates the Hold

Test Case Iteration	Results
Grandstream Budgetone 100 Series	Pass
Grandstream GXP-2000	Pass

Action: Make a voice call from vendor SIP Phone to Cisco SIP Phone.

Action: Answer the call

Verification(1): Talk path is available.

Action: Vendor SIP phone places the call on Hold.

Verification(2): Holding party user is placed on hold.

Action: user Resume the call.

Verification(3): Talk path is available.

Action: vendor SIP Phone clears the call.

Verification(4): The call is cleared

Result: passed?___X_____failed?_________

Comments:
When the vendor SIP phone places the call on hold, the last digit of the calling number displayed on the Cisco SIP phone is missing. This is ccm issue CSCsc96611.

7.1.2 Vendor SIP to Cisco SIP Voice call – terminating end initiates the Hold

Test Case Iteration	Results
Grandstream Budgetone 100 Series	Pass
Grandstream GXP-2000	Pass

Action: Make a voice call from vendor SIP Phone to Cisco SIP Phone.

Action: Answer the call.

Verification(1): Talk path is available.

Action: Cisco SIP Phone puts the call on Hold.

Verification(2): Vendor SIP Phone is placed on hold.

Action: Cisco SIP Phone resumes the call.

Verification(3): Talk path is available.

Action: vendor SIP Phone clears the call.

Verification(4): The call is cleared

Result: passed?___X_____failed?________

Comments:
When the Cisco SIP Phone places the call on hold, the last digit of the calling number displayed on the Cisco SIP phone is missing. This is ccm issue CSCsc96611.

7.1.3 Vendor SIP to Cisco SCCP Voice call – originating end initiates the Hold

Test Case Iteration	Results
Grandstream Budgetone 100 Series	Pass
Grandstream GXP-2000	Pass

Action: Make a voice call from vendor SIP Phone to Cisco IP Phone.

Action: Answer the call.

Verification(1): Talk path is available.

Action: Vendor SIP Phone puts the call on Hold.

Verification(2): Cisco SCCP IP Phone user is placed on hold.

Action: vendor SIP Phone user Resume the call.

Verification(3): Talk path is available.

Action: vendor SIP Phone user clears the call.

Verification(4): The call is cleared.

Result: *passed?___X_____failed?________*

Comments:
When the vendor SIP phone places the call on hold, the last digit of the calling number displayed on Cisco SCCP phone is missing. This is ccm issue CSCsc96611.

7.1.4 Vendor SIP to Cisco SCCP Voice call – terminating end initiates the Hold

Test Case Iteration	Results
Grandstream Budgetone 100 Series	Pass
Grandstream GXP-2000	Pass

Action: Make a voice call from vendor SIP Phone to Cisco SCCP IP Phone

Action: Answer the call at Cisco SCCP IP Phone

Verification(1): Talk path is available.

Action: Cisco SCCP IP Phone user puts the call on Hold.

Verification(2): vendor SIP Phone user is placed on hold.

Action: Cisco SCCP IP Phone resumes the call.

Verification(3): Talk path is available.

Action: vendor SIP Phone user clears the call.

Verification(4): The call is cleared

Result: *passed?___X____failed?________*

Comments:
When the Cisco SCCP Phone places the call on hold, the last digit of the calling number displayed on Cisco SCCP phone is missing. This is ccm issue CSCsc96611.

7.1.5 Cisco SCCP to vendor SIP Voice call – originating end initiates the Hold

Test Case Iteration	Results
Grandstream Budgetone 100 Series	Pass
Grandstream GXP-2000	Pass

Action: Make a voice call from Cisco SCCP IP Phone to vendor SIP Phone

Action: Answer the call at vendor SIP Phone

Verification(1): Talk path is available.

Action: Cisco SCCP IP Phone user puts the call on Hold.

Verification(2): vendor SIP Phone user is placed on hold.

Action: Cisco SCCP IP Phone user Resume the call.

Verification(3): Talk path is available.

Action: Cisco SCCP IP Phone user clears the call.

Verification(4): The call is cleared

Result: *passed?__X______failed?________*

Comments:

7.1.6 Cisco SCCP to vendor SIP Voice call – terminating end initiates the Hold

Test Case Iteration	Results
Grandstream Budgetone 100 Series	Pass
Grandstream GXP-2000	Pass

Action: Make a voice call from Cisco SCCP IP Phone to vendor SIP Phone

Action: Answer the call at vendor SIP Phone

Verification(1): Talk path is available.

Action: vendor SIP Phone user puts the call on Hold.

Verification(2): Cisco SCCP IP Phone user is placed on hold.

Action: vendor SIP Phone user Resume the call.

Verification(3): Talk path is available.

Action: Cisco SCCP IP Phone user clears the call.

Verification(4): The call is cleared

Result: *passed?*___*X*_____*failed?*________

Comments:

8 Call Transfer

8.1 Description

Call Transfer (CT) is a supplementary service which enables a served user (User A) to transform two of that users calls into a new call between the other two users of the two calls (User B and User C). Each call can either be an incoming call to User A or an outgoing call from User A. After successful invocation of CT, User B and User C will no longer be able to communicate with User A.

In a Call Transfer, there are three actors. The person being transferred is known as the transferee. The person transferring the call is known as the transferor. The person receiving the transfer is known as the transfer target or simply the target. There are three Call Transfer types. Attended, Early Attended, and Blind.

8.1.1 Attended

With attended transfer, the transferor places the transferee on hold and calls the target. After conversing with the target, the transferor completes the transfer and drops out of both calls. The transferee is automatically taken off of hold and connected to the target.

8.1.2 Early Attended

With early attended transfer, the transferor places the original call on hold and calls the target. Upon hearing ringback tone, the transferor transfers the call to the target and drops out of both calls. The transferee hears ring back while the target's phone is alerting. When the target answers, a connection is established between transferee and target.

8.1.3 Blind

With blind transfer, the transferor places the original call on hold and dials the target. The transferor then uses SIP signaling to redirect the transferee to the target. No call is made to the target prior to transfer. The timing of when the transferor drops out of the call depends on the transferor's implementation of the feature, but most likely the drop occurs when the transferor is notified that the redirect operation was accepted and has begun.

8.2 Call Transfer Test Cases

8.2.1 Vendor SIP to vendor SIP Attended Call Transfer completes – secondary call answered

Test Case Iteration	Results
Grandstream Budgetone 100 Series	Pass, see comments
Grandstream GXP-2000	Pass, see comments

Action: vendor SIP Phone calls first Cisco SIP Phone. Call is answered.

Action: vendor SIP Phone initiates transfer and calls second Vendor SIP Phone. Second Vendor SIP Phone answers the call.

Action: vendor SIP Phone completes transfer (Manufacturer dependent).

Verification (1): Verify that the Cisco SIP Phone and vendor SIP Phone are connected, and that vendor SIP Phone is disconnected from call.

Result: passed?____X____ failed?_________

Comments:
In order to support Attended Call Transfer on the Grandstream (both BT100 and GXP-2000) phones, the "MTP Required" check box needs to be checked in the CCM's Phone configuration for that particular phone. Furthermore the Attended transfer should be done from the same line number (for GXP-2000 with multi-line configuration) on the Grandstream GXP-2000 phone since CCM does not support transferring a call on one line to a call on another line. The calls have to be on the same line.

To do Attended Transfer from Grandstream Budgetone 100 phone the user presses the "FLASH" button and hears a dial tone, then dial the phone number followed by pressing the "SEND" button. If the call is answered the user presses "TRANSFER" to complete the transfer operation.

To do Attended Transfer from Grandstream GXP-2000 phone the user needs to put the active line on hold by pressing the "HOLD" button. User makes a second call using same line by pressing hookflash (if using the handset) or "SPEAKER" button twice (if in speaker mode) to get dial tone on same line. User will then press "TRNF" button, then press the intended line that was put on hold.

8.2.2 Vendor SIP to Cisco SCCP Attended Call Transfer completes – secondary call answered

Test Case Iteration	Results
Grandstream Budgetone 100 Series	Pass, see comments
Grandstream GXP-2000	Pass, see comments

Action: vendor SIP Phone calls Cisco IP Phone (SCCP). Cisco IP Phone answers call.

Action: vendor SIP Phone initiates transfer to second Cisco IP Phone. Call is answered.

Action: vendor SIP Phone completes transfer (Manufacturer dependent).

Verification (1): Verify that the two Cisco IP Phones are connected, and vendor SIP Phone is disconnected from the call.

Result: passed?___X_____ failed?_________

Comments:
In order to support Attended Call Transfer on the Grandstream (both BT100 and GXP-2000) phones, the "MTP Required" check box needs to be checked in the CCM's Phone configuration for that particular phone. Furthermore the Attended transfer should be done from the same line number (for GXP-2000 with multi-line configuration) on the Grandstream GXP-2000 phone since CCM does not support transferring a call on one line to a call on another line. The calls have to be on the same line.

8.2.3 Vendor SIP to Cisco SIP Attended Call Transfer completes – secondary call answered

Test Case Iteration	Results
Grandstream Budgetone 100 Series	Pass, see comments
Grandstream GXP-2000	Pass, see comments

Action: vendor SIP Phone calls Cisco SIP Phone. Cisco SIP Phone answers call.

Action: vendor SIP Phone initiates transfer to second Cisco SIP Phone. Call is answered.

Action: vendor SIP Phone completes transfer (Manufacturer dependent).

Verification (1): Verify that the two Cisco SIP Phones are connected, and vendor SIP Phone is disconnected from the call.

Result: *passed?___X_____ failed?_________*

Comments:
In order to support Attended Call Transfer on the Grandstream (both BT100 and GXP-2000) phones, the "MTP Required" check box needs to be checked in the CCM's Phone configuration for that particular phone. Furthermore the Attended transfer should be done from the same line number (for GXP-2000 with multi-line configuration) on the Grandstream GXP-2000 phone since CCM does not support transferring a call on one line to a call on another line. The calls have to be on the same line.

8.2.4 Cisco SCCP to vendor SIP Attended Call Transfer completes – secondary call answered

Test Case Iteration	Results
Grandstream Budgetone 100 Series	Pass
Grandstream GXP-2000	Pass

Action: Cisco IP Phone (SCCP) calls vendor SIP Phone. Call is answered.

Action: Cisco IP Phone initiates transfer and calls second Vendor SIP Phone. Call is answered.

Action: Cisco IP Phone completes transfer.

Verification (1): Verify that vendor SIP Phones are connected, and Cisco IP Phone is disconnected from call.

Result: *passed?____X____ failed?_________*

Comments:

8.2.5 Vendor SIP to vendor SIP Early Attended Call Transfer completes – secondary call answered

Test Case Iteration	Results
Grandstream Budgetone 100 Series	No Supported
Grandstream GXP-2000	No Supported

Action: vendor SIP Phone calls first Cisco SIP Phone. Call is answered.

Action: vendor SIP Phone initiates transfer and calls second Vendor SIP Phone. Ringback tone is heard on (transfering) vendor SIP Phone.

Action: vendor SIP Phone completes transfer.

Verification (1): Verify that the Cisco SIP phone hears ringback tone.

Action: second vendor SIP Phone answers the call.

Verification (2): Verify that the Cisco SIP and Vendor SIP Phones are connected, and that vendor SIP Phone is disconnected from call.

Result: *passed?___X_____* *failed?_________*

Comments:
Grandstream phones did not support Early Attended Call Transfer.

8.2.6 Vendor SIP to Cisco SIP Early Attended Call Transfer completes – secondary call answered

Test Case Iteration	Results
Grandstream Budgetone 100 Series	No Supported
Grandstream GXP-2000	No Supported

Action: vendor SIP Phone calls first Cisco SIP Phone. Call is answered.

Action: vendor SIP Phone initiates transfer and calls second Cisco SIP Phone. Ringback tone is heard on (transfering) vendor SIP Phone.

Action: vendor SIP Phone completes transfer.

Verification (1): Verify that the first Cisco SIP phone hears ringback tone.

Action: second Cisco SIP Phone answers the call.

Verification (2): Verify that the Cisco SIP Phones are connected, and that vendor SIP Phone is disconnected from call.

Result: *passed?*___X_____ *failed?*_________

Comments:
Grandstream phones did not support Early Attended Call Transfer.

8.2.7 Vendor SIP to Cisco SCCP Early Attended Call Transfer completes – secondary call answered

Test Case Iteration	Results
Grandstream Budgetone 100 Series	No Supported
Grandstream GXP-2000	No Supported

Action: vendor SIP Phone calls Cisco IP Phone (SCCP). Cisco IP Phone answers call.

Action: vendor SIP Phone initiates transfer to second Cisco IP Phone. Ringback tone is heard on (transfering) vendor SIP Phone.

Action: vendor SIP Phone completes transfer.

Verification (1): Verify that the first Cisco SIP phone hears ringback tone.

Action: second Cisco SIP Phone answers the call.

Verification (2): Verify that the two Cisco IP Phones are connected, and vendor SIP Phone is disconnected from the call.

Result: *passed?*________ *failed?*_________

Comments:
Grandstream phones did not support Early Attended Call Transfer.

8.2.8 Cisco SCCP to vendor SIP Early Attended Call Transfer completes – secondary call answered

Test Case Iteration	Results
Grandstream Budgetone 100 Series	Pass
Grandstream GXP-2000	Pass

Action: Cisco IP Phone (SCCP) calls vendor SIP Phone. Call is answered.

Action: Cisco IP Phone initiates transfer and calls second Vendor SIP Phone. Ringback tone is heard on (transfering) Cisco IP Phone.

Action: Cisco IP Phone completes transfer.

Verification (1): Verify that the first vendor SIP phone hears ringback tone.

Action: second vendor SIP Phone answers the call.

Verification (2): Verify that vendor SIP Phones are connected, and Cisco IP Phone is disconnected from call.

Result: *passed?___X_____* *failed?__________*

Comments:

8.2.9 Vendor SIP to vendor SIP Blind Call Transfer completes – call answered

Test Case Iteration	Results
Grandstream Budgetone 100 Series	Pass
Grandstream GXP-2000	Pass

Action: vendor SIP Phone calls first Cisco SIP Phone. Call is answered.

Action: vendor SIP Phone initiates blind transfer (Manufacturer dependent) and calls second Vendor SIP Phone.
Verification (1): Verify that ringback tone is heard at first Cisco SIP phone, and vendor SIP Phone is dropped from call.

Action: Answer call at second Vendor SIP Phone.

Verification (2): Verify that Cisco SIP Phone and Vendor SIP Phone are connected, and that vendor SIP Phone is disconnected from call.

Result: *passed?*___X_____ *failed?*_________

Comments:
To do Blind Transfer from Grandstream BT100 phone the user presses the "TRANSFER" button and hears a dial tone, then dial the phone number followed by pressing the "SEND" button. At this point the user can hangup. This will transfer the other party to the dialed number.

To do Blind Transfer from Grandstream GXP-2000 phone the user presses the "TRNF" button and hears a dial tone, then dial the phone number followed by pressing the "SEND" button. At this point the user can hangup. This will transfer the other party to the dialed number.

8.2.10 Vendor SIP to Cisco SCCP Blind Call Transfer completes – call answered

Test Case Iteration	Results
Grandstream Budgetone 100 Series	Pass
Grandstream GXP-2000	Pass

Action: vendor SIP Phone calls Cisco IP Phone (SCCP). Cisco IP Phone answers call.

Action: vendor SIP Phone initiates Blind transfer to second Cisco IP Phone.

Verification (1): Verify that ringback tone is heard at first Cisco IP phone, and vendor SIP phone is dropped from call.

Action: Answer call at second Cisco IP Phone.

Verification (2): the two Cisco IP Phones are connected, and that vendor SIP Phone is disconnected from call.

Result: *passed?*____X____ *failed?*_________

Comments:

8.2.11 Cisco SCCP to Vendor SIP Blind Call Transfer completes – call answered

Test Case Iteration	Results
Grandstream Budgetone 100 Series	Not Applicable
Grandstream GXP-2000	Not Applicable

Action: Cisco IP Phone (SCCP) calls vendor SIP Phone. Call is answered

Action: Cisco IP Phone initiates Blind transfer to second Vendor IP Phone.

Verification (1): Verify that ringback tone is heard at first Cisco IP phone, and vendor SIP phone is dropped from call.

Action: Answer call at second Vendor IP Phone.

Verification (2): Verify that Cisco IP Phone and Vendor SIP Phone are connected, and that vendor SIP Phone is disconnected from call.

Result: *passed?*________ *failed?*_________

Comments:
Cisco SCCP IP Phone does not support Blind Call Transfer.

9 Call Diversion (CFU-Call Forwarding Unconditional), (CFB-Call Forwarding Busy), (CFNA-Call Forwarding No Answer)

9.1 Description

Diversion enables a served user to have the telephone set redirect calls to other numbers. The served user's ability to originate calls is unaffected by Diversion.
After CFU has been activated calls are forwarded independently of the status of the served user.
After CFB has been activated calls are forwarded if the served user is busy.
After CFNA has been activated calls are forwarded for which the connection is not established within a predefined period of time.

9.2 Test configuration for Diversion

When testing Call Diversion (Forwarding), vendor and Cisco SIP Phones, as well as Cisco SCCP IP Phones are registered to CallManager 5.0. Call Forwarding busy/no answer/all calls is manually invoked at the vendor SIP Phone via feature button or access codes.

9.3 Test Cases

These tests should be done only if both vendors support Diversion and can have it running on all interfaces. Calling Number/Name and Connection Number/Name should be turned on for all the following cases unless otherwise specified.

9.3.1 Vendor SIP to Cisco SIP Call Forwarding

Test Case Iteration	CF-TYPE	Results
Grandstream Budgetone 100 Series	Unconditional	Pass
	Busy	Pass
	No Answer	Pass
Grandstream GXP-2000	Unconditional	Pass
	Busy	Pass, see note [1]
	No Answer	Pass

Action: vendor SIP Phone invokes Call Forwarding to Cisco SIP Phone.

Action: From another phone on CallManager (SCCP or SIP), call the forwarded SIP Phone.

Verification (1): Verify that the call is forwarded to forwarded-to Cisco SIP Phone.

Verification (2): Verify that the calling phone displays forwarding station's name and number forwarding to forwarded-to Cisco SIP Phone's name and number.

Verification (3): Verify that forwarded-to Cisco SIP Phone display shows calling station name and number calling forwarding station's name and number with forward indication.

Action: Answer the call from forwarded to Cisco SIP Phone.

Verification (4): The talk path between calling station and forwarded-to station is established.

Verification (5): Calling station display shows forwarding station's name and number forwarding to forwarded-to station's name and number.

[1] To trigger CFB from the GXP-2000 phone instead of CCM, we have to set the busy trigger for the CCM's DN for that particular phone higher than the busy trigger on the Grandstream phone which is 11 calls.

Verification (6): The display of forwarded-to station shows calling station's name and number calling forwarding station's name and number with forward indication.

Action: Detail the displays and then hang-up.

Result: *passed?*_________*failed?*____*X*____

Comments:

For Grandstream BT100 and GXP-2000 Phones:
Call Forward Unconditional: to use this feature the user dials "*72" and get the dial tone. Then dial the forward number followed by "#" button, wait for dial tone then hangup. To cancel CFU dial "*73" and get the dial tone then hangup.

Call Forward Busy: to use this feature the user dials "*90" and get the dial tone. Then dial the forward number followed by "#" button, wait for dial tone then hangup. To cancel CFB dial "*91" and get the dial tone then hangup.

Call Forward No Answer (Delayed Call Forward): to use this feature the user dials "*92" and get the dial tone. Then dial the forward number followed by "#" button, wait for dial tone then hangup. To cancel CFB dial "*93" and get the dial tone then hangup.

Verification(2): Calling phone displays forwarded-to Cisco SIP Phone's Alerting Name and number.
Verification(3): forwarded-to Cisco SIP Phone display shows calling station name and number calling forwarding station's name and number with forward indication.
Verification(5): Calling phone displays forwarded-to Cisco SIP Phone's Name and number.
Verification(6): forwarded-to Cisco SIP Phone display shows calling station's name and number.

9.3.2 Vendor SIP to Vendor SIP Call Forwarding

Test Case Iteration	CF-TYPE	Results
Grandstream Budgetone 100 Series	Unconditional	Pass
	Busy	Pass
	No Answer	Pass
Grandstream GXP-2000	Unconditional	Pass
	Busy	Pass
	No Answer	Pass

Action: vendor SIP Phone invokes Call Forwarding to Vendor SIP Phone.

Action: From another phone on CallManager (SCCP or SIP), call the forwarded SIP Phone.

Verification (1): Verify that the call is forwarded to forwarded-to Vendor SIP Phone.

Verification (2): Verify that the calling phone displays forwarding station's name and number forwarding to forwarded-to Vendor SIP Phone's name and number.

Verification (3): Verify that forwarded-to Vendor SIP Phone display shows calling station name and number calling forwarding station's name and number with forward indication.

Action: Answer the call from forwarded to Vendor SIP Phone.

Verification (4): The talk path between calling station and forwarded-to station is established.

Verification (5): Calling station display shows forwarding station's name and number forwarding to forwarded-to station's name and number.

Verification (6): The display of forwarded-to station shows calling station's name and number calling forwarding station's name and number with forward indication.

Action: Detail the displays and then hang-up.

Result: *passed?*_________*failed?*___*X*_____

Comments:

For Grandstream BT100 Phone:
Verification(2): Calling phone displays forwarded-to vendor SIP Phone's Alerting Name and Number.
Verification(3): forwarded-to vedor SIP Phone display shows calling station's Number.
Verification(5): Calling phone displays forwarded-to vendor SIP Phone's Name and Number.
Verification(6): forwarded-to vendor SIP Phone display shows calling station's Number.
For Grandstream GXP-2000 Phone:
Verification(2): Calling phone displays forwarded-to vendor SIP Phone's Alerting Name and number.
Verification(3): forwarded-to vedor SIP Phone display shows calling station's Name and Number.
Verification(5): Calling phone displays forwarded-to vendor SIP Phone's Name and Number.
Verification(6): forwarded-to vendor SIP Phone display shows calling station's Name and Number.

9.3.3 Vendor SIP to Cisco SCCP Call Forwarding

Test Case Iteration	CF-TYPE	Results
Grandstream Budgetone 100 Series	Unconditional	Pass
	Busy	Pass
	No Answer	Pass
Grandstream GXP-2000	Unconditional	Pass
	Busy	Pass

	No Answer	Pass

Action: vendor SIP Phone invokes Call Forwarding to Cisco SCCP IP Phone.

Action: From another phone on CallManager (SCCP or SIP), call the forwarded SIP Phone.

Verification (1): Verify that the call is forwarded to forwarded-to Cisco SCCP IP Phone.

Verification (2): Verify that the calling phone displays the forwarding station's name and number forwarding to forwarded-to Cisco IP Phone's name and number.

Verification (3): Verify that forwarded-to Cisco IP Phone display shows calling station name and number calling forwarding station's name and number with forward indication.

Action: Answer the call from forwarded-to Cisco IP Phone.

Verification (4): The talk path between calling station and forwarded-to station is established.

Verification (5): Calling station display shows forwarding station's name and number forwarding to forwarded-to station's name and number.

Verification (6): The display of forwarded-to station shows calling station's name and number calling forwarding station's name and number with forward indication.

Action: Detail the displays and then hang-up.

Result: *passed?*_________*failed?*___*X*_____

Comments:
For Grandstream BT100 and GXP-2000 Phones:

CFU:
Verification(2): Calling phone displays forwarded-to Cisco SCCP Phone's Alerting Name and number.
Verification(3): forwarded-to Cisco SCCP Phone display shows calling station name and number calling forwarding station's name and number with forward indication.
Verification(5): Calling phone displays forwarded-to Cisco SCCP Phone's Name and number.
Verification(6): forwarded-to Cisco SCCP Phone display shows calling station's name and number.

CFB, CFNR:
Verification(2): Calling phone displays forwarded-to Cisco SCCP Phone's Alerting Name and number.
Verification(3): forwarded-to Cisco SCCP Phone display shows calling station's name and number.
Verification(5): Calling phone displays forwarded-to Cisco SCCP Phone's Name and number.
Verification(6): forwarded-to Cisco SCCP Phone display shows calling station's name and number.

10 Call Conference

10.1 Description

Call Conference enables a served user to add a conferee to the call

10.2 Test Configuration for Conference

When testing Call Conference, vendor and Cisco SIP Phones, as well as Cisco IP Phones (SCCP) and Cisco Conference bridge(s) are registered to CallManager 5.0

10.3 Call Conference Test Cases

10.3.1 Vendor SIP to Vendor SIP Call Conference

Test Case Iteration	Results
Grandstream Budgetone 100 Series	Pass
Grandstream GXP-2000	Pass

Action: vendor SIP Phone calls Cisco SIP Phone. Call is answered.

Action: vendor SIP Phone initiates call conference (Manufacturer dependent) and calls second Vendor SIP Phone. Call is answered at second Vendor SIP Phone.

Action: vendor SIP Phone completes the conference (Manufacturer dependent).

Verification (1): Verify that both vendor SIP Phones are connected to Cisco SIP Phones.

Verification (2): Verify voice path between all three stations involved in conference.

Action: vendor SIP Phone goes on-hook and hangs up the call (Manufacturer dependent).

Verification (3): Verify that the Cisco SIP phone stays connected to second Vendor SIP Phone.

Optional Verification (4): (name and/or number display is optional) verify that stations' displays are updated when conference is established and after conference call is dropped.

Result: *passed?*________ *failed?*___X______

Comments:
For Grandstream BT100 Phone:
To do 3-way Call Conference from Grandstream BT100 phone the user presses "CONFERENCE" button to get dial tone and put the active call on hold. The user then dials the number then presses "SEND"

button to make the call. When the destination phone answers the call, the user presses "CONFERENCE" button again to join Calling and Called party into the conference.

For Grandstream GXP-2000 Phone:
To do 3-way Call Conference from Grandstream GXP-2000 phone the user needs to put the active line on hold by pressing the "HOLD" button. User makes a second call by pressing another line button to get dial tone. User will then dial the number then presses "SEND" to make the second call. When the destination phone answers the call, the user presses "CONF" button then presses the line that is on hold to join Calling and Called party into the conference.

10.3.2 Vendor SIP to Cisco SIP Call Conference

Test Case Iteration	Results
Grandstream Budgetone 100 Series	Pass
Grandstream GXP-2000	Pass

Action: vendor SIP Phone calls Cisco SIP Phone. Call is answered.

Action: vendor SIP Phone initiates call conference (Manufacturer dependent) and calls second Cisco SIP Phone. Call is answered at second Cisco SIP Phone.

Action: vendor SIP Phone completes the conference (Manufacturer dependent).

Verification (1): Verify that vendor SIP Phone is connected to both Cisco SIP Phones.

Verification (2): Verify voice path between all three stations involved in conference.

Action: vendor SIP Phone goes on-hook and hangs up the call (Manufacturer dependent).

Verification (3): Verify that first Cisco SIP phone stays connected to second Cisco SIP Phone.

Optional Verification (4): (name and/or number display is optional) verify that station's displays are updated when conference is established and after conference call is dropped.

Result: *passed?________* *failed?___X______*

Comments:

10.3.3 Vendor SIP to Cisco SCCP Call Conference

Test Case Iteration	Results
Grandstream Budgetone 100 Series	Pass
Grandstream GXP-2000	Pass

Action: vendor SIP Phone calls Cisco IP Phone (SCCP). Call is answered.

Action: vendor SIP Phone initiates call conference (Manufacturer dependent) and calls second IP Phone. Call is answered at second Cisco IP Phone.

Action: vendor SIP Phone completes the conference (Manufacturer dependent).

Verification (1): Verify that vendor SIP Phone is connected to both Cisco IP Phones.

Verification (2): Verify voice path between all three stations involved in conference.

Action: vendor SIP Phone goes on-hook and hangs up the call (Manufacturer dependent).

Verification (3): Verify that first Cisco IP phone stays connected to second Cisco IP Phone.

Optional Verification (4): (name and/or number display is optional) verify that stations' displays are updated when conference is established and after conference call is dropped.

Result: *passed?*_________ *failed?*___*X*______

Comments:

10.3.4 Cisco SCCP to Vendor SIP Call Conference

Test Case Iteration	Results
Grandstream Budgetone 100 Series	Pass
Grandstream GXP-2000	Pass

Action: Cisco IP Phone (SCCP) calls vendor SIP Phone. Call is answered.

Action: Cisco IP Phone (SCCP) initiates call conference and calls second Vendor SIP Phone. Call is answered at second Vendor SIP Phone.

Action: Cisco IP Phone completes the conference.

Verification (1): Verify that vendor SIP Phones are connected to Cisco IP Phones.

Verification (2): Verify voice path between all three stations involved in conference.

Action: Cisco IP Phone goes on-hook and hangs up the call.

Verification (3): Verify that second Vendor SIP phone stays connected to first vendor SIP Phone.

Optional Verification (4): (name and/or number display is optional) verify that stations' displays are updated when conference is established and after conference call is dropped.

Result: passed?___X_____ failed?_________

Comments:
For Grandstream Phone:
Optional Verification(4): Display of vendor SIP phones are not upated after conferencing phone is dropped.

11 Voicemail access and MWI support.

11.1 Description

When accessing voicemail, SIP Phones must be able to generate and pass DTMF tones to voicemail servers. Also, SIP Phones should be able to support message waiting indication updates via display notification and/or MWI.

11.2 Test configuration for Voicemail and MWI support

When testing voicemail access and MWI, vendor SIP Phones, as well as Cisco Phones are registered on same CallManager 5.0 (cluster A). Cisco Unity Ports are registered on a different CallManager 5.0 (cluster B).

11.3 Test Cases

11.3.1 Generic voicemail access/ DTMF tone outpulsing to Cisco Unity ports

Test Case Iteration	Results
Grandstream Budgetone 100 Series	Pass
Grandstream GXP-2000	Pass

Action: vendor SIP Phone calls Cisco Unity pilot number. Upon answer, access voicemail mailbox by pressing proper digits on phone keypad.

Verification (1): Cisco Unity properly responds and interprets digits being passed by SIP Phone.

Result: *passed?______failed?___X_____*

Comments:

11.3.2 Message Waiting Indication – MWI on

Test Case Iteration	Results
Grandstream Budgetone 100 Series	Pass
Grandstream GXP-2000	Pass

Action: Leave new voicemail message into Cisco Unity mailbox assigned to vendor SIP Phone

Verification (1): vendor SIP Phone message waiting indicator is activated.

Result: *passed?__X___failed?________*

Comments:

11.3.3 Message Waiting Indication – MWI off

Test Case Iteration	Results
Grandstream Budgetone 100 Series	Pass
Grandstream GXP-2000	Pass

Action: Access Unity mailbox and listen/delete new voicemail message

Verification (1): vendor SIP Phone message waiting indicator is deactivated.

Result: *passed?__X____failed?________*

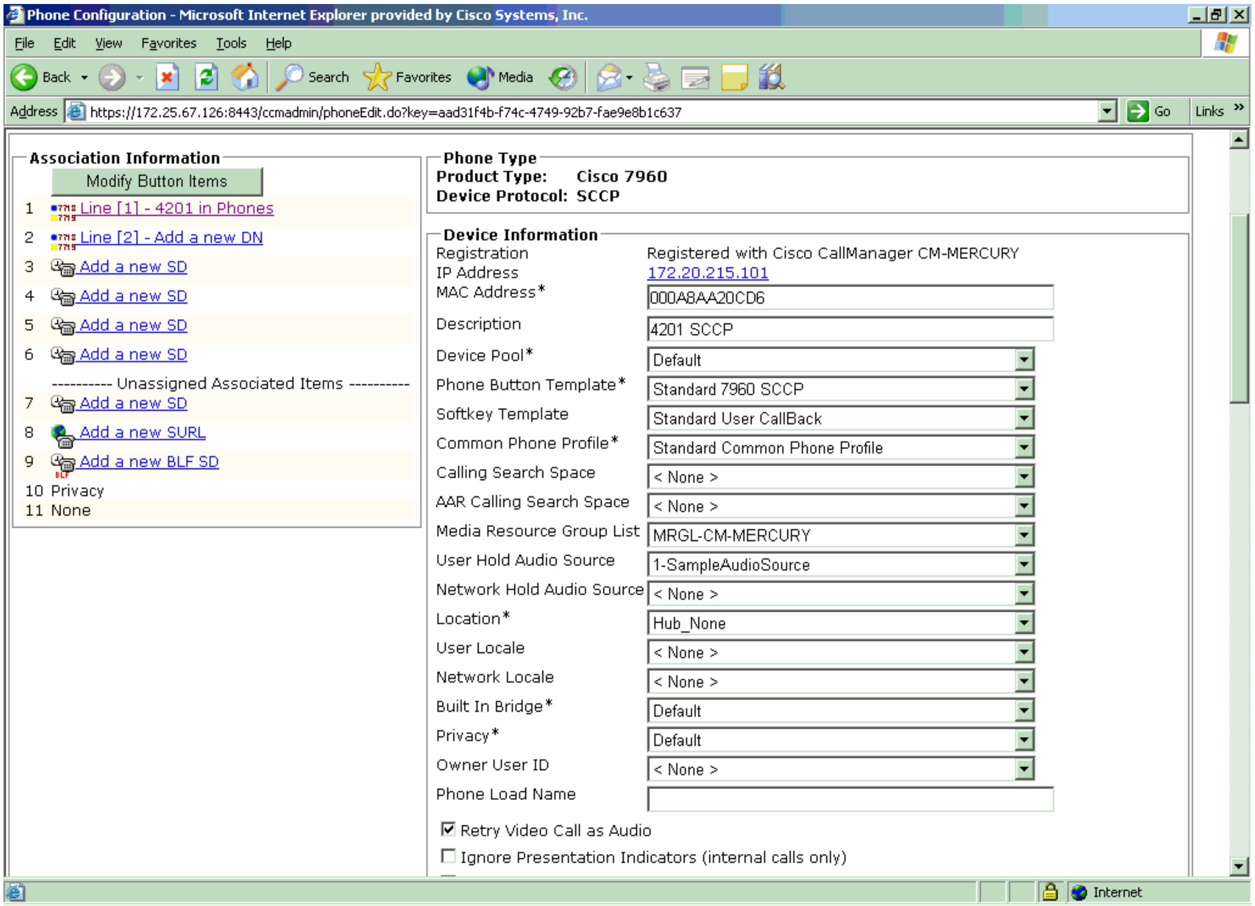
Comments:

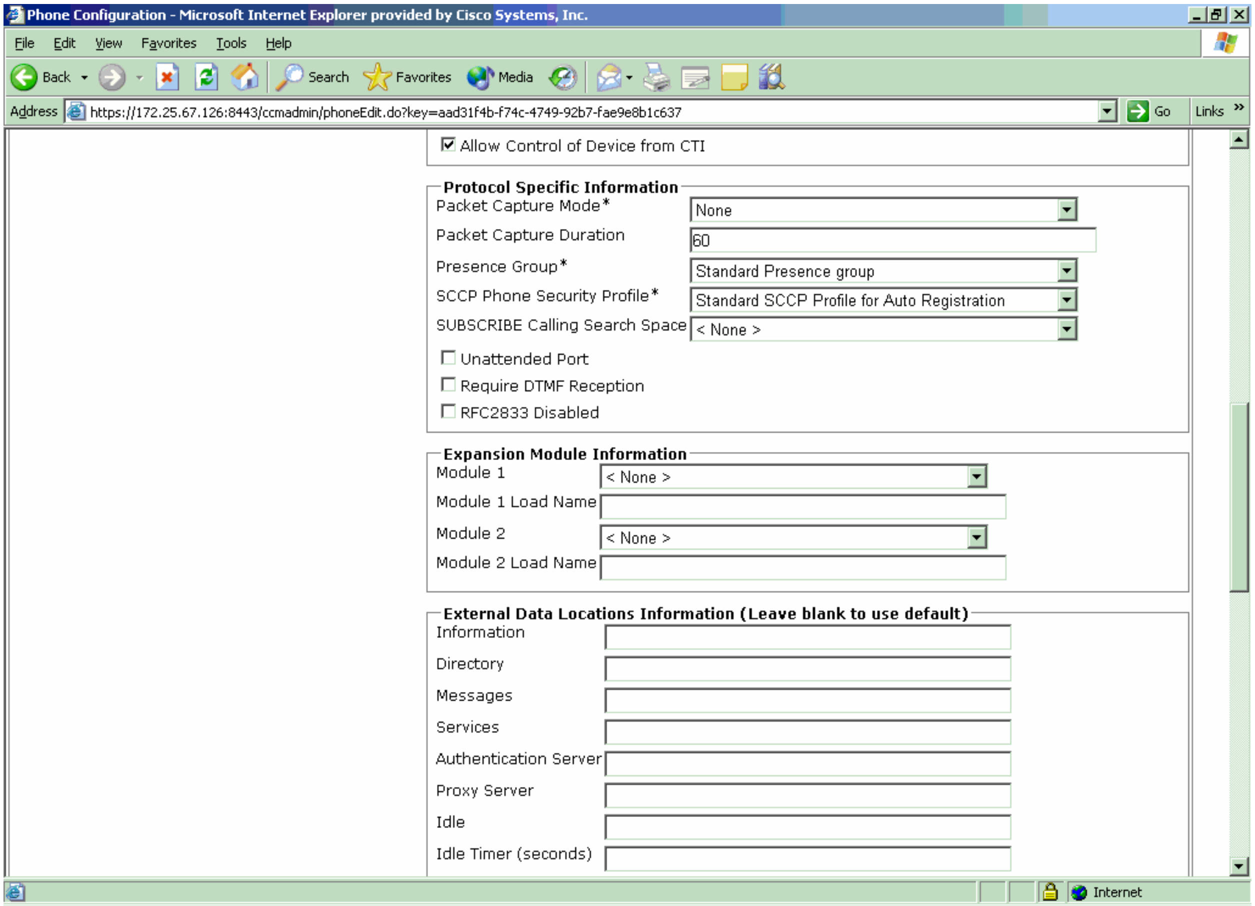
12 Product Configurations

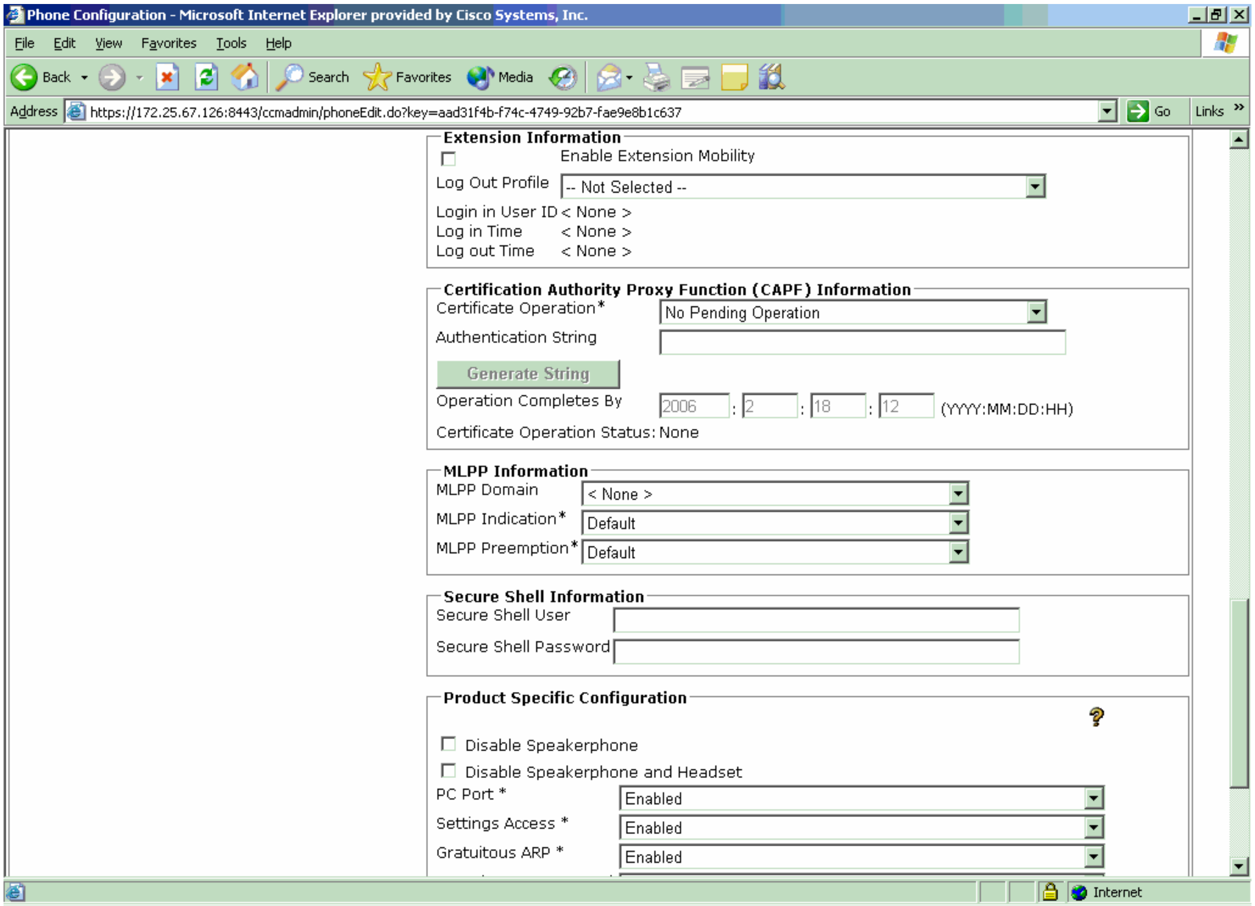
12.1 Cisco CCM configuration

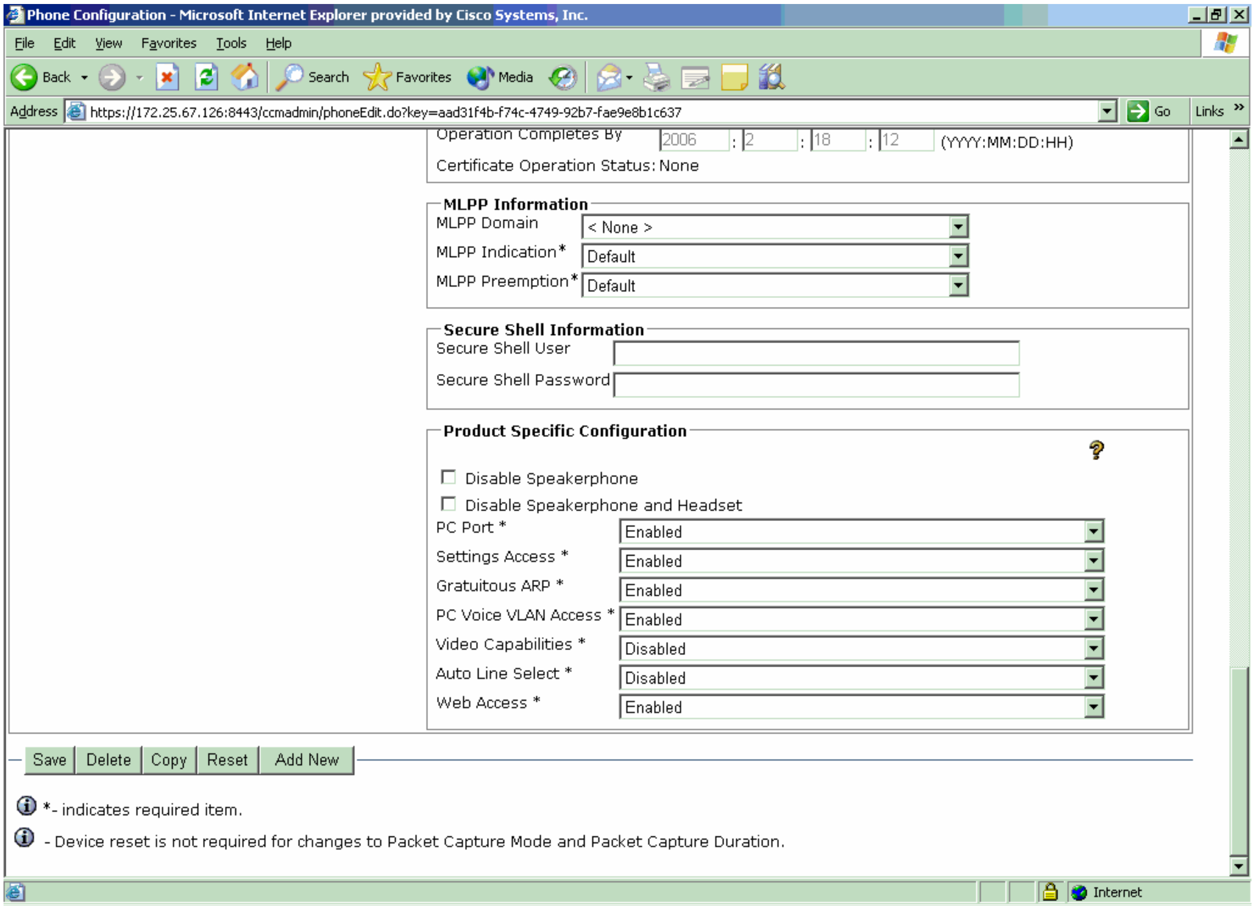
12.1.1 CCM Version

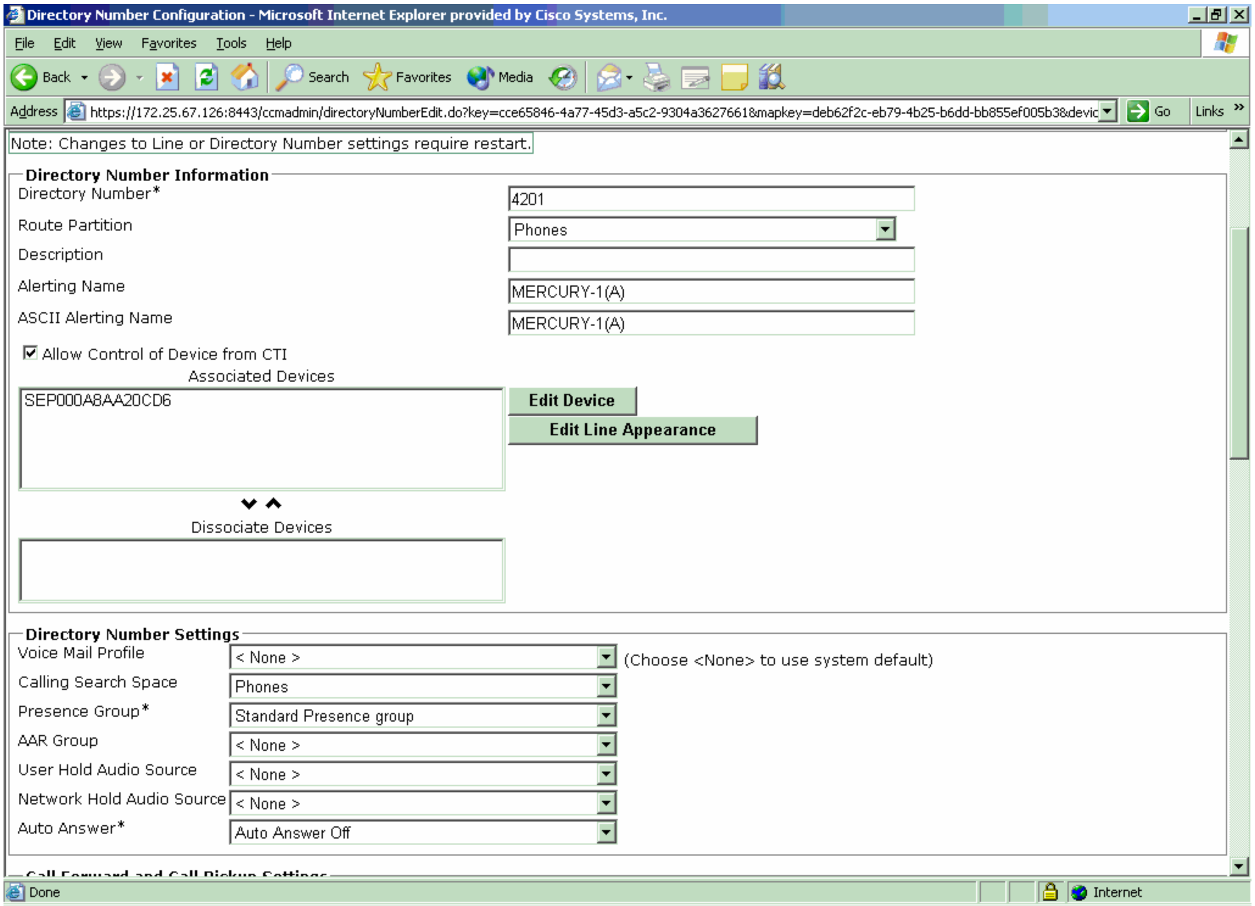
12.1.2 Cisco 7960 Phone Configurations

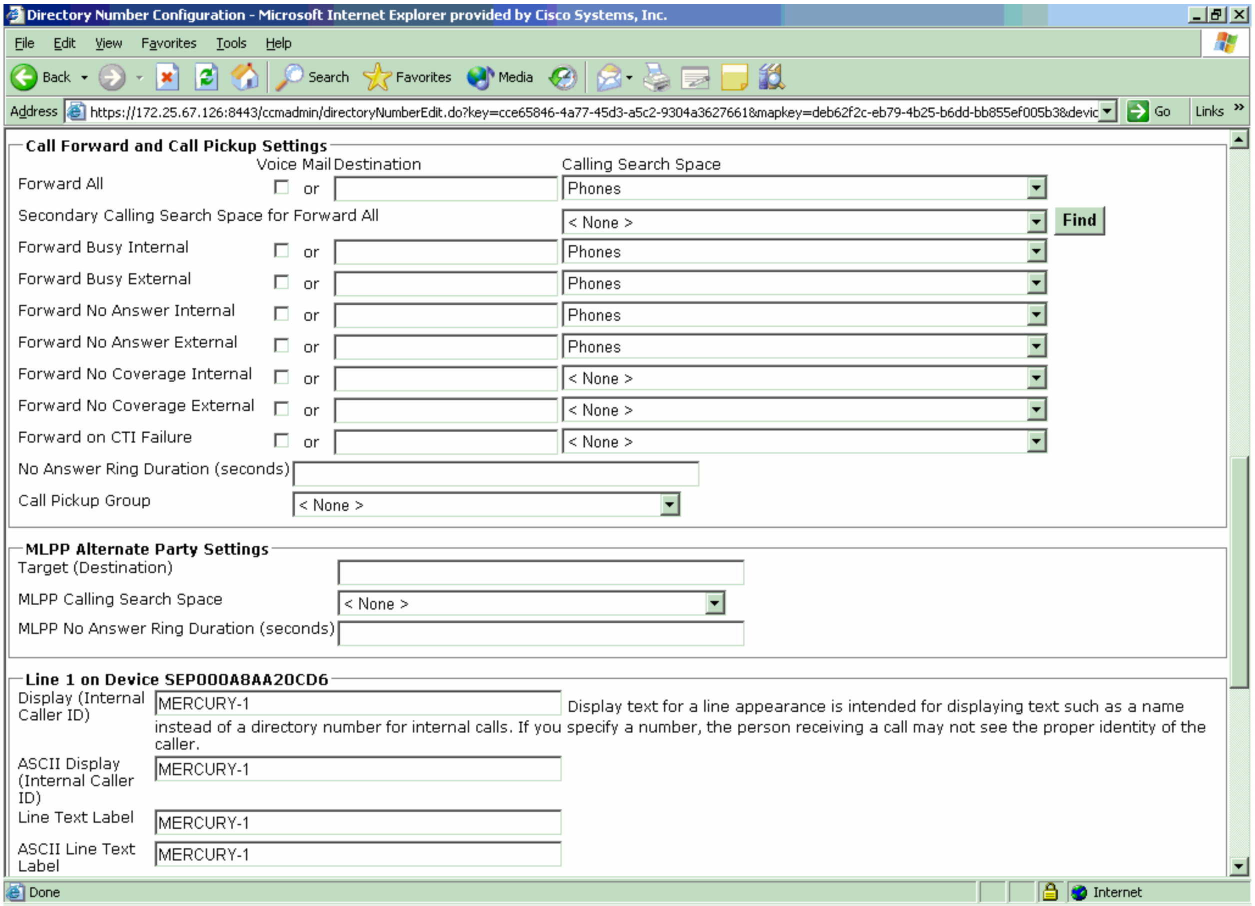



Phone Configuration - Microsoft Internet Explorer provided by Cisco Systems, Inc.
File Edit View Favorites Tools Help
Back Search Favorites Media
Address https://172.25.67.126:8443/ccmadmin/phoneEdit.do?key=aad31f4b-f74c-4749-92b7-fae9e8b1c637
Go Links
Allow Control of Device from CTI
Protocol Specific Information
Packet Capture Mode* None
Packet Capture Duration 60
Presence Group* Standard Presence group
SCCP Phone Security Profile* Standard SCCP Profile for Auto Registration
SUBSCRIBE Calling Search Space < None >
Unattended Port
Require DTMF Reception
RFC2833 Disabled
Expansion Module Information
Module 1 < None >
Module 1 Load Name
Module 2 < None >
Module 2 Load Name
External Data Locations Information (Leave blank to use default)
Information
Directory
Messages
Services
Authentication Server
Proxy Server
Idle
Idle Timer (seconds)
Internet

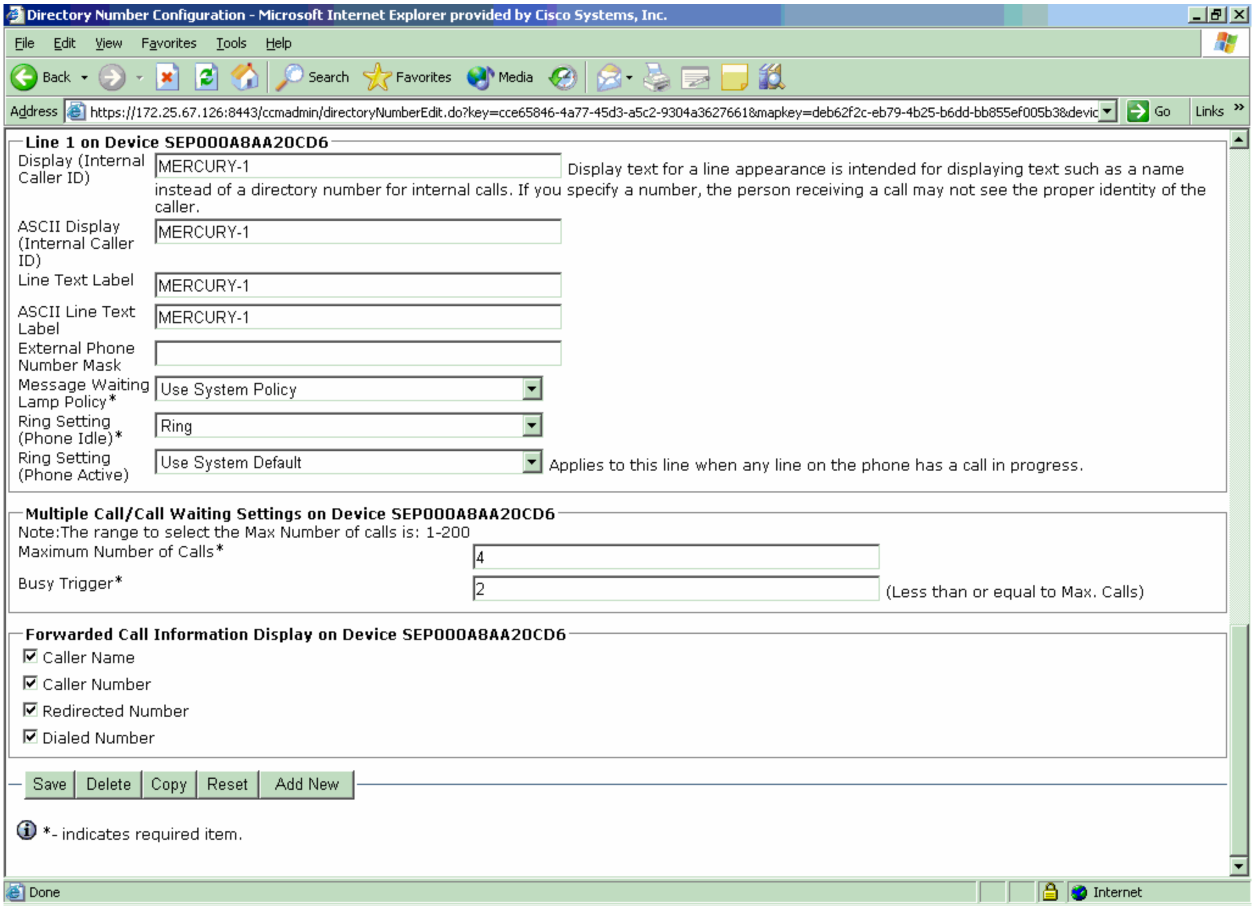

Phone Configuration - Microsoft Internet Explorer provided by Cisco Systems, Inc.
File Edit View Favorites Tools Help
Back Search Favorites Media
Address https://172.25.67.126:8443/ccmadmin/phoneEdit.do?key=aad31f4b-f74c-4749-92b7-fae9e8b1c637
Go Links
Extension Information
Enable Extension Mobility
Log Out Profile -- Not Selected --
Login in User ID < None >
Log in Time < None >
Log out Time < None >
Certification Authority Proxy Function (CAPF) Information
Certificate Operation* No Pending Operation
Authentication String
Generate String
Operation Completes By 2006 : 2 : 18 : 12 (YYYY:MM:DD:HH)
Certificate Operation Status: None
MLPP Information
MLPP Domain < None >
MLPP Indication* Default
MLPP Preemption* Default
Secure Shell Information
Secure Shell User
Secure Shell Password
Product Specific Configuration
Disable Speakerphone
Disable Speakerphone and Headset
PC Port * Enabled
Settings Access * Enabled
Gratuitous ARP * Enabled
Internet


Phone Configuration - Microsoft Internet Explorer provided by Cisco Systems, Inc.
File Edit View Favorites Tools Help
Back Search Favorites Media
Address https://172.25.67.126:8443/ccmadmin/phoneEdit.do?key=aad31f4b-f74c-4749-92b7-fae9e8b1c637
Go Links
Operation Completes By 2006 : 2 : 18 : 12 (YYYY:MM:DD:HH)
Certificate Operation Status: None
MLPP Information
MLPP Domain < None >
MLPP Indication* Default
MLPP Preemption* Default
Secure Shell Information
Secure Shell User
Secure Shell Password
Product Specific Configuration
Disable Speakerphone
Disable Speakerphone and Headset
PC Port * Enabled
Settings Access * Enabled
Gratuitous ARP * Enabled
PC Voice VLAN Access * Enabled
Video Capabilities * Disabled
Auto Line Select * Disabled
Web Access * Enabled
Save Delete Copy Reset Add New
*- indicates required item.
- Device reset is not required for changes to Packet Capture Mode and Packet Capture Duration.
Internet

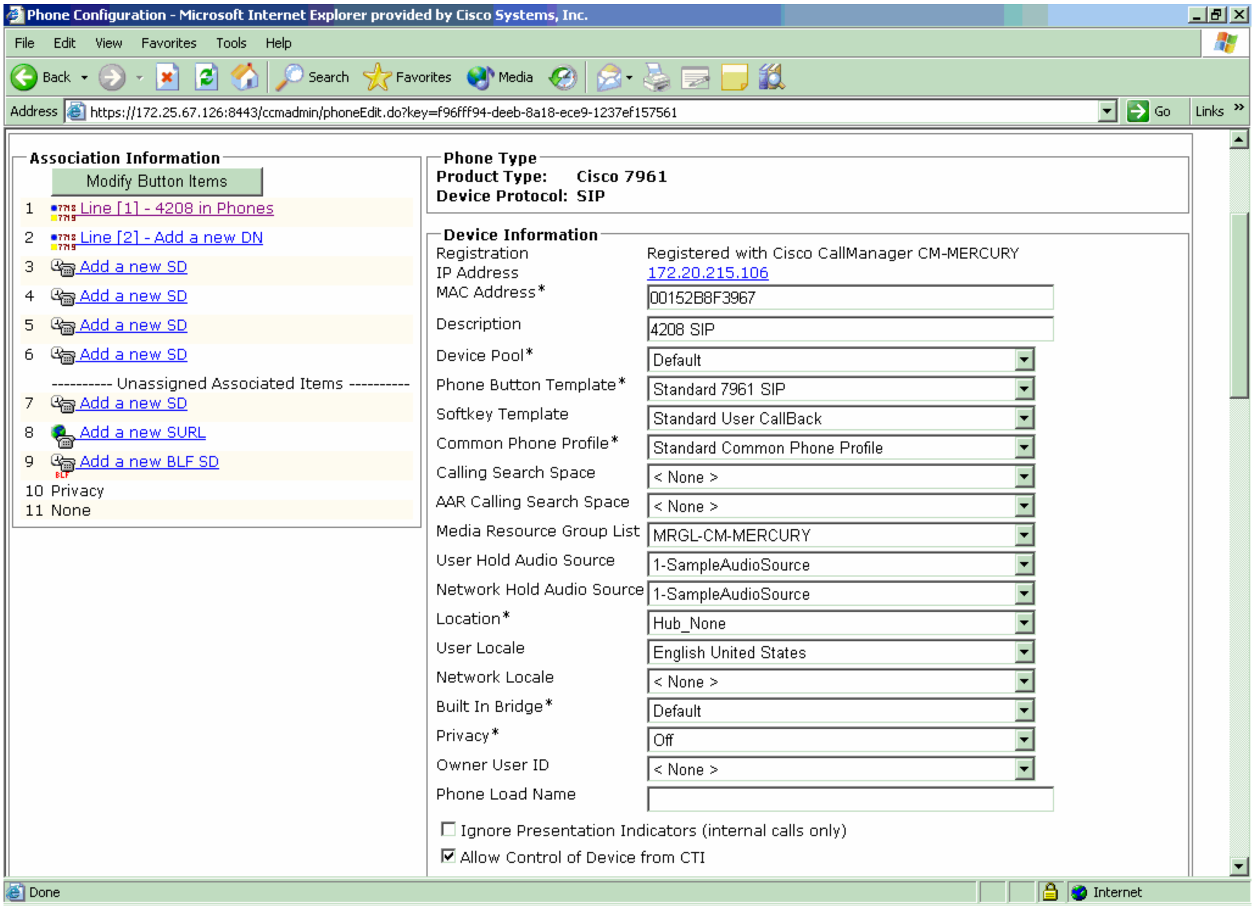

Directory Number Configuration - Microsoft Internet Explorer provided by Cisco Systems, Inc.
File Edit View Favorites Tools Help
Back Search Favorites Media
Address https://172.25.67.126:8443/ccmadmin/directoryNumberEdit.do?key=cce65846-4a77-45d3-a5c2-9304a3627661&mapkey=deb62f2c-eb79-4b25-b6dd-bb855ef005b3&devic
Go Links
Note: Changes to Line or Directory Number settings require restart.
Directory Number Information
Directory Number*
4201
Route Partition
Phones
Description
Alerting Name
MERCURY-1(A)
ASCII Alerting Name
MERCURY-1(A)
Allow Control of Device from CTI
Associated Devices
SEP000A8AA20CD6
Edit Device
Edit Line Appearance
Dissociate Devices
Directory Number Settings
Voice Mail Profile
< None >
(Choose <None> to use system default)
Calling Search Space
Phones
Presence Group*
Standard Presence group
AAR Group
< None >
User Hold Audio Source
< None >
Network Hold Audio Source
< None >
Auto Answer*
Auto Answer Off
Done
Internet

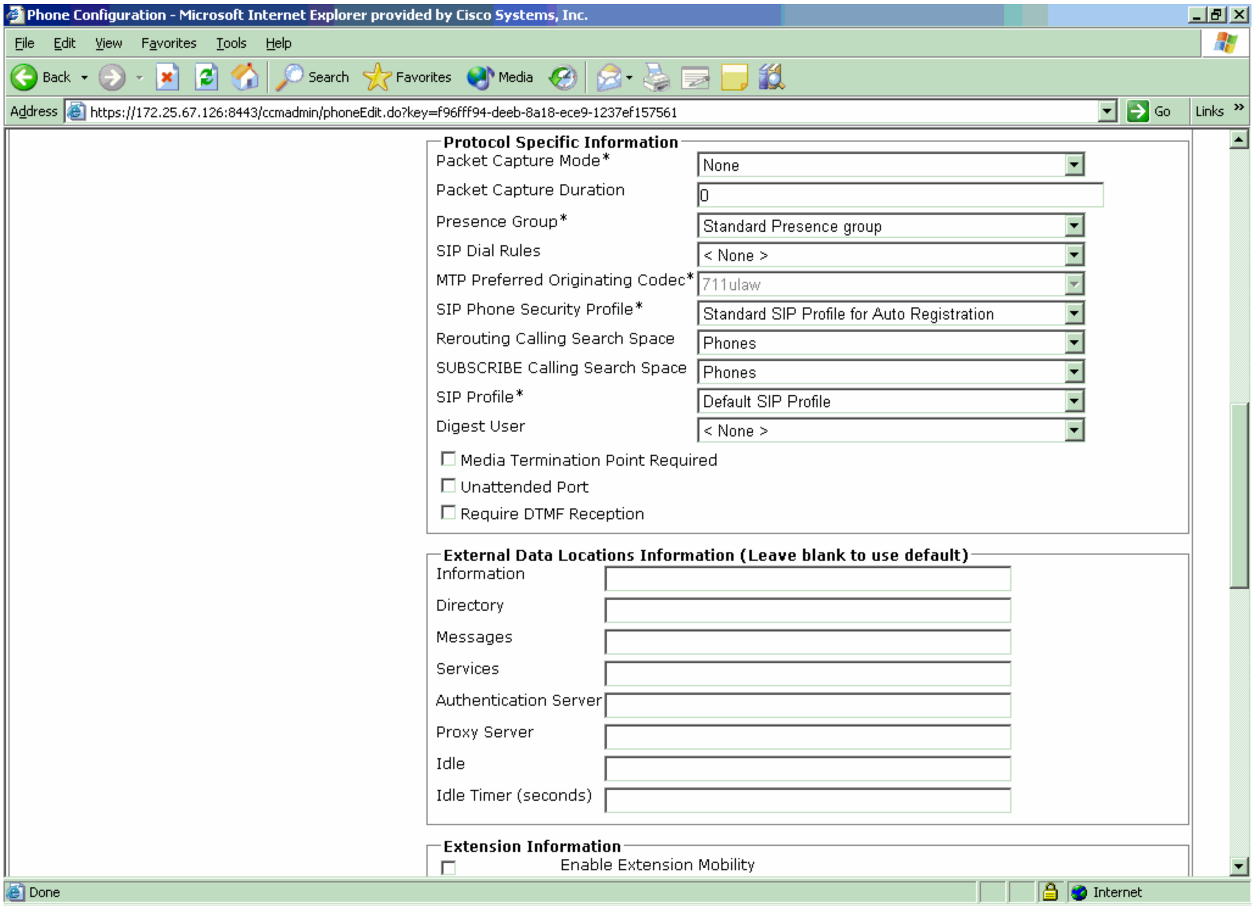


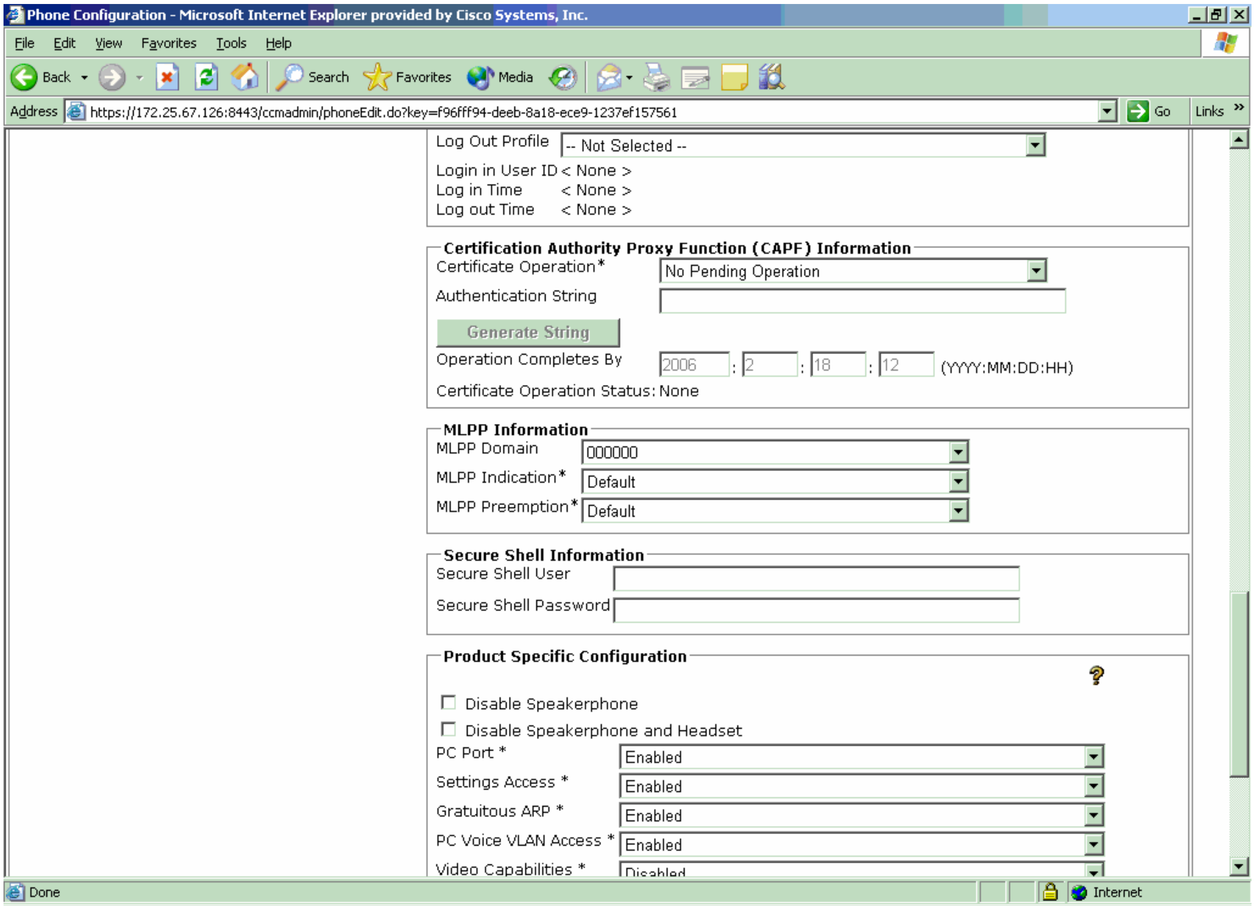
Directory Number Configuration - Microsoft Internet Explorer provided by Cisco Systems, Inc.
File Edit View Favorites Tools Help
Back Search Favorites Media
Address https://172.25.67.126:8443/ccmadmin/directoryNumberEdit.do?key=cce65846-4a77-45d3-a5c2-9304a3627661&mapkey=deb62f2c-eb79-4b25-b6dd-bb855ef005b3&devic
Go Links
Call Forward and Call Pickup Settings
Voice Mail Destination
Calling Search Space
Forward All or Phones
Secondary Calling Search Space for Forward All < None > Find
Forward Busy Internal or Phones
Forward Busy External or Phones
Forward No Answer Internal or Phones
Forward No Answer External or Phones
Forward No Coverage Internal or < None >
Forward No Coverage External or < None >
Forward on CTI Failure or < None >
No Answer Ring Duration (seconds)
Call Pickup Group < None >
MLPP Alternate Party Settings
Target (Destination)
MLPP Calling Search Space < None >
MLPP No Answer Ring Duration (seconds)
Line 1 on Device SEP000A8AA20CD6
Display (Internal Caller ID) MERCURY-1
Display text for a line appearance is intended for displaying text such as a name instead of a directory number for internal calls. If you specify a number, the person receiving a call may not see the proper identity of the caller.
ASCII Display (Internal Caller ID) MERCURY-1
Line Text Label MERCURY-1
ASCII Line Text Label MERCURY-1
Done
Internet

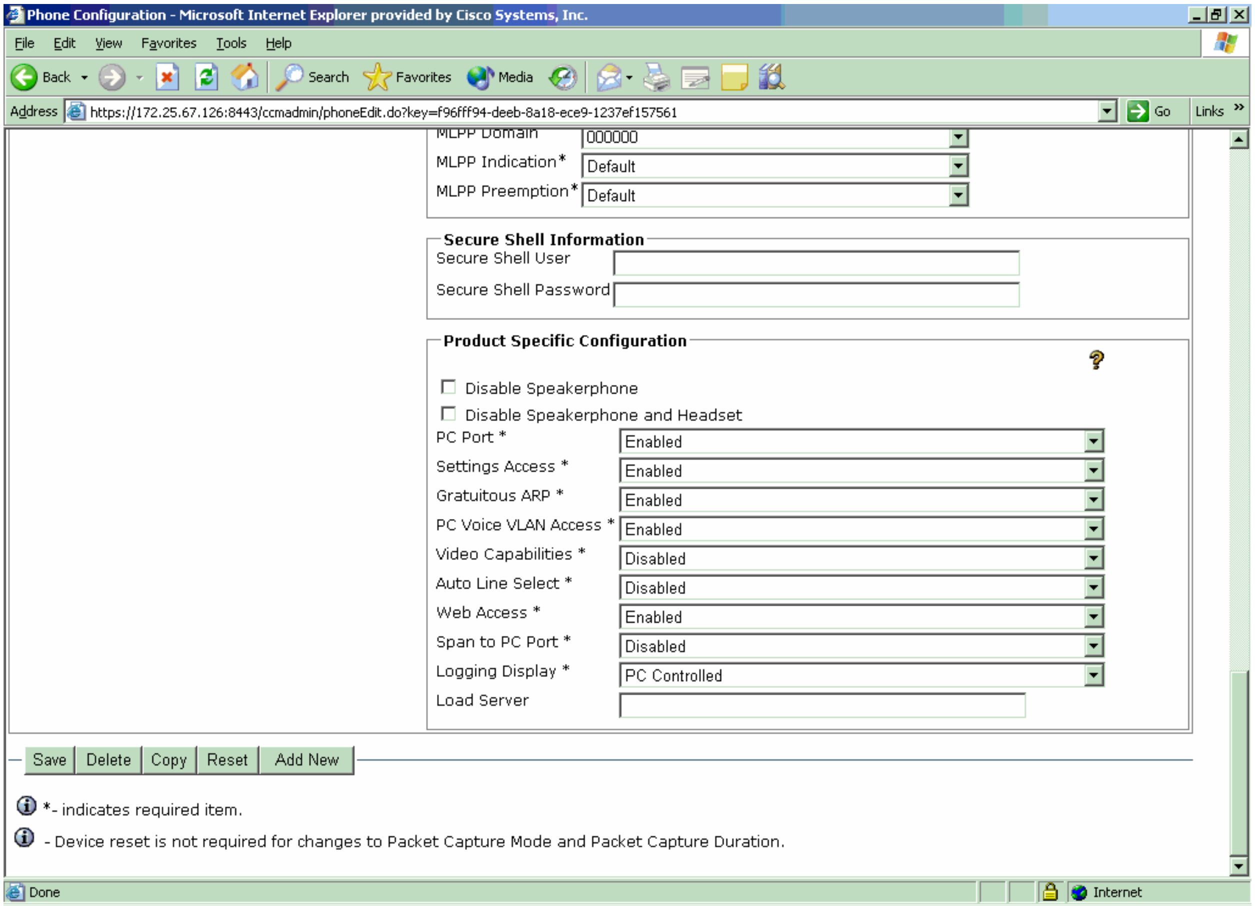

Directory Number Configuration - Microsoft Internet Explorer provided by Cisco Systems, Inc.
File Edit View Favorites Tools Help
Back Search Favorites Media
Address https://172.25.67.126:8443/ccmadmin/directoryNumberEdit.do?key=cce65846-4a77-45d3-a5c2-9304a3627661&mapkey=deb62f2c-eb79-4b25-b6dd-bb855ef005b3&devic
Go Links
Line 1 on Device SEP000A8AA20CD6
Display (Internal Caller ID)
MERCURY-1
Display text for a line appearance is intended for displaying text such as a name instead of a directory number for internal calls. If you specify a number, the person receiving a call may not see the proper identity of the caller.
ASCII Display (Internal Caller ID)
MERCURY-1
Line Text Label
MERCURY-1
ASCII Line Text Label
MERCURY-1
External Phone Number Mask
Message Waiting Lamp Policy*
Use System Policy
Ring Setting (Phone Idle)*
Ring
Ring Setting (Phone Active)
Use System Default
Applies to this line when any line on the phone has a call in progress.
Multiple Call/Call Waiting Settings on Device SEP000A8AA20CD6
Note:The range to select the Max Number of calls is: 1-200
Maximum Number of Calls*
4
Busy Trigger*
2
(Less than or equal to Max. Calls)
Forwarded Call Information Display on Device SEP000A8AA20CD6
Caller Name
Caller Number
Redirected Number
Dialed Number
Save
Delete
Copy
Reset
Add New
*- indicates required item.
Done
Internet

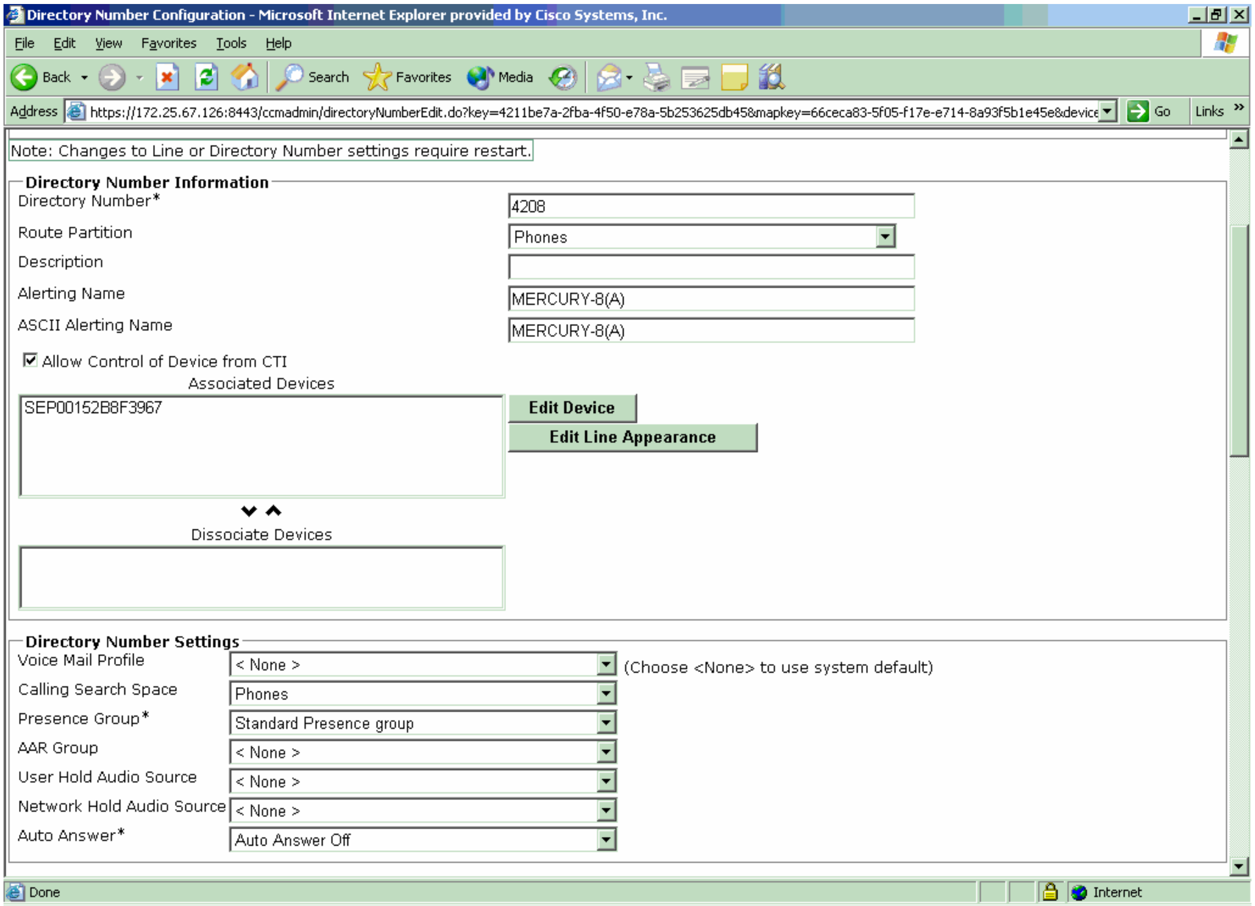
12.1.3 Cisco 7961 SIP Phone Configurations

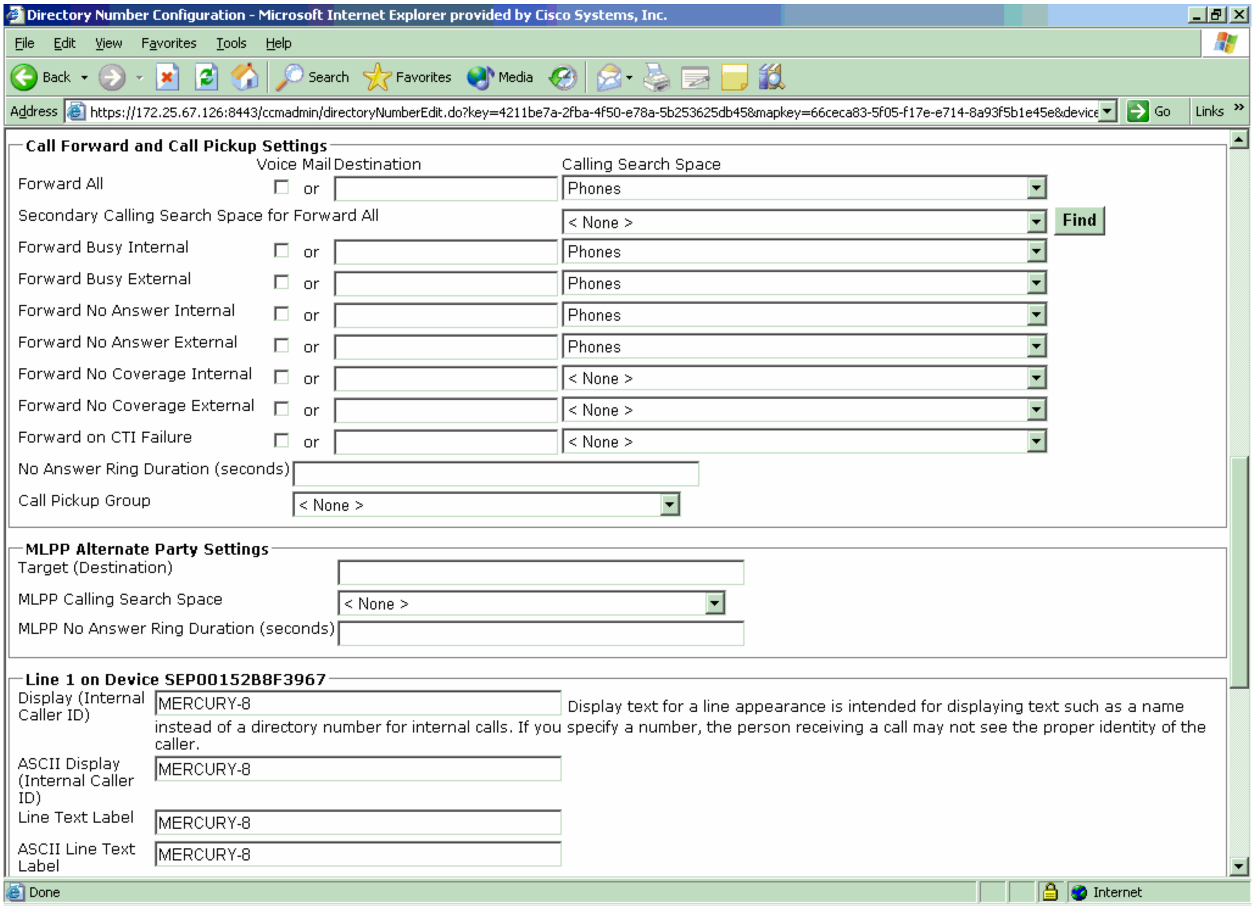


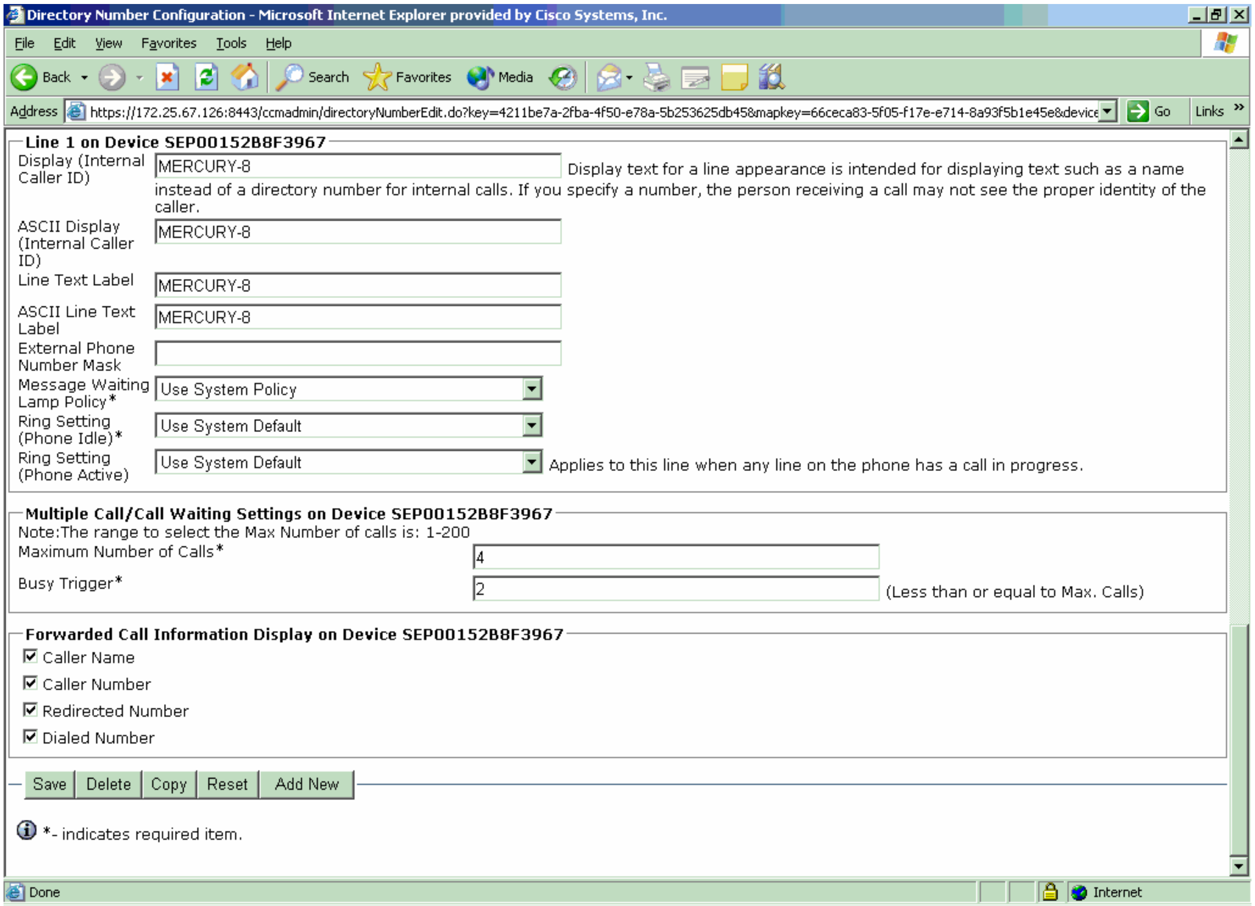

Phone Configuration - Microsoft Internet Explorer provided by Cisco Systems, Inc.
File Edit View Favorites Tools Help
Back Search Favorites Media
Address https://172.25.67.126:8443/ccmadmin/phoneEdit.do?key=f96fff94-deeb-8a18-ece9-1237ef157561
Go Links
Protocol Specific Information
Packet Capture Mode* None
Packet Capture Duration 0
Presence Group* Standard Presence group
SIP Dial Rules < None >
MTP Preferred Originating Codec* 711ulaw
SIP Phone Security Profile* Standard SIP Profile for Auto Registration
Rerouting Calling Search Space Phones
SUBSCRIBE Calling Search Space Phones
SIP Profile* Default SIP Profile
Digest User < None >
Media Termination Point Required
Unattended Port
Require DTMF Reception
External Data Locations Information (Leave blank to use default)
Information
Directory
Messages
Services
Authentication Server
Proxy Server
Idle
Idle Timer (seconds)
Extension Information
Enable Extension Mobility
Done
Internet


Phone Configuration - Microsoft Internet Explorer provided by Cisco Systems, Inc.
File Edit View Favorites Tools Help
Back Search Favorites Media
Address https://172.25.67.126:8443/ccmadmin/phoneEdit.do?key=f96fff94-deeb-8a18-ece9-1237ef157561
Go Links
Log Out Profile -- Not Selected --
Login in User ID < None >
Log in Time < None >
Log out Time < None >
Certification Authority Proxy Function (CAPF) Information
Certificate Operation* No Pending Operation
Authentication String
Generate String
Operation Completes By 2006 : 2 : 18 : 12 (YYYY:MM:DD:HH)
Certificate Operation Status: None
MLPP Information
MLPP Domain 000000
MLPP Indication* Default
MLPP Preemption* Default
Secure Shell Information
Secure Shell User
Secure Shell Password
Product Specific Configuration
Disable Speakerphone
Disable Speakerphone and Headset
PC Port * Enabled
Settings Access * Enabled
Gratuitous ARP * Enabled
PC Voice VLAN Access * Enabled
Video Capabilities *
Done
Internet


Phone Configuration - Microsoft Internet Explorer provided by Cisco Systems, Inc.
File Edit View Favorites Tools Help
Back Search Favorites Media
Address https://172.25.67.126:8443/ccmadmin/phoneEdit.do?key=f96fff94-deeb-8a18-ece9-1237ef157561
Go Links
MLPP Domain 000000
MLPP Indication* Default
MLPP Preemption* Default
Secure Shell Information
Secure Shell User
Secure Shell Password
Product Specific Configuration
Disable Speakerphone
Disable Speakerphone and Headset
PC Port * Enabled
Settings Access * Enabled
Gratuitous ARP * Enabled
PC Voice VLAN Access * Enabled
Video Capabilities * Disabled
Auto Line Select * Disabled
Web Access * Enabled
Span to PC Port * Disabled
Logging Display * PC Controlled
Load Server
Save Delete Copy Reset Add New
*- indicates required item.
- Device reset is not required for changes to Packet Capture Mode and Packet Capture Duration.
Done
Internet


Directory Number Configuration - Microsoft Internet Explorer provided by Cisco Systems, Inc.
File Edit View Favorites Tools Help
Back Search Favorites Media
Address https://172.25.67.126:8443/ccmadmin/directoryNumberEdit.do?key=4211be7a-2fba-4f50-e78a-5b253625db45&mapkey=66ceca83-5f05-f17e-e714-8a93f5b1e45e&device
Go Links
Note: Changes to Line or Directory Number settings require restart.
Directory Number Information
Directory Number* 4208
Route Partition Phones
Description
Alerting Name MERCURY-8(A)
ASCII Alerting Name MERCURY-8(A)
Allow Control of Device from CTI
Associated Devices
SEP00152B8F3967
Edit Device
Edit Line Appearance
Dissociate Devices
Directory Number Settings
Voice Mail Profile < None > (Choose <None> to use system default)
Calling Search Space Phones
Presence Group* Standard Presence group
AAR Group < None >
User Hold Audio Source < None >
Network Hold Audio Source < None >
Auto Answer* Auto Answer Off
Done
Internet


Directory Number Configuration - Microsoft Internet Explorer provided by Cisco Systems, Inc.
File Edit View Favorites Tools Help
Back Search Favorites Media
Address https://172.25.67.126:8443/ccmadmin/directoryNumberEdit.do?key=4211be7a-2fba-4f50-e78a-5b253625db45&mapkey=66ceca83-5f05-f17e-e714-8a93f5b1e45e&device
Go Links
Call Forward and Call Pickup Settings
Voice Mail Destination
Calling Search Space
Forward All or Phones
Secondary Calling Search Space for Forward All < None > Find
Forward Busy Internal or Phones
Forward Busy External or Phones
Forward No Answer Internal or Phones
Forward No Answer External or Phones
Forward No Coverage Internal or < None >
Forward No Coverage External or < None >
Forward on CTI Failure or < None >
No Answer Ring Duration (seconds)
Call Pickup Group < None >
MLPP Alternate Party Settings
Target (Destination)
MLPP Calling Search Space < None >
MLPP No Answer Ring Duration (seconds)
Line 1 on Device SEP00152B8F3967
Display (Internal Caller ID) MERCURY-8
Display text for a line appearance is intended for displaying text such as a name instead of a directory number for internal calls. If you specify a number, the person receiving a call may not see the proper identity of the caller.
ASCII Display (Internal Caller ID) MERCURY-8
Line Text Label MERCURY-8
ASCII Line Text Label MERCURY-8
Done
Internet



Directory Number Configuration - Microsoft Internet Explorer provided by Cisco Systems, Inc.

File Edit View Favorites Tools Help

Back Search Favorites Media

Address https://172.25.67.126:8443/ccmadmin/directoryNumberEdit.do?key=4211be7a-2fba-4f50-e78a-5b253625db45&mapkey=66ceca83-5f05-f17e-e714-8a93f5b1e45e&device Go Links

Line 1 on Device SEP00152B8F3967

Display (Internal Caller ID)	MERCURY-8	Display text for a line appearance is intended for displaying text such as a name instead of a directory number for internal calls. If you specify a number, the person receiving a call may not see the proper identity of the caller.
ASCII Display (Internal Caller ID)	MERCURY-8	
Line Text Label	MERCURY-8	
ASCII Line Text Label	MERCURY-8	
External Phone Number Mask		
Message Waiting Lamp Policy*	Use System Policy	
Ring Setting (Phone Idle)*	Use System Default	
Ring Setting (Phone Active)	Use System Default	Applies to this line when any line on the phone has a call in progress.

Multiple Call/Call Waiting Settings on Device SEP00152B8F3967

Note:The range to select the Max Number of calls is: 1-200

Maximum Number of Calls*	4	
Busy Trigger*	2	(Less than or equal to Max. Calls)

Forwarded Call Information Display on Device SEP00152B8F3967

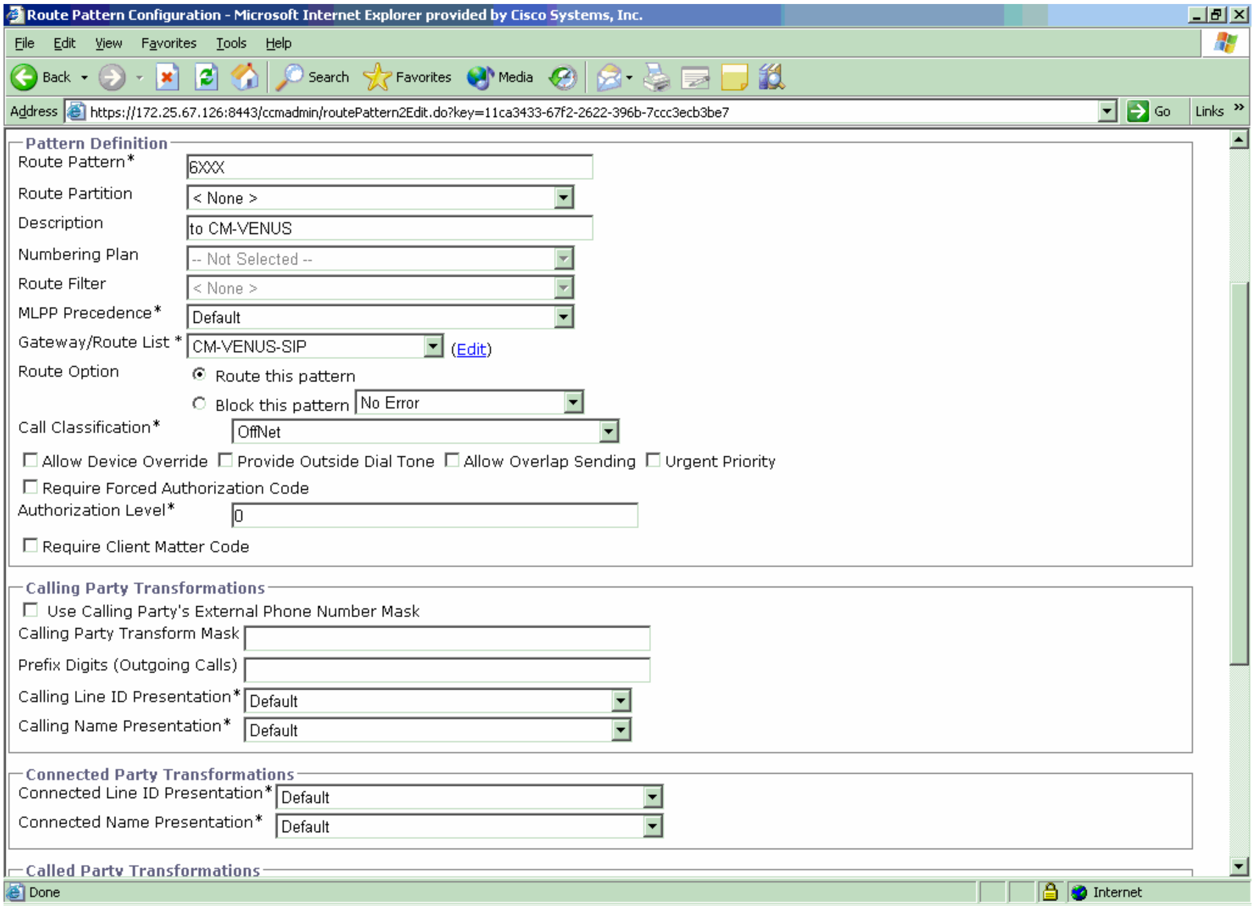
- [x] Caller Name
- [x] Caller Number
- [x] Redirected Number
- [x] Dialed Number

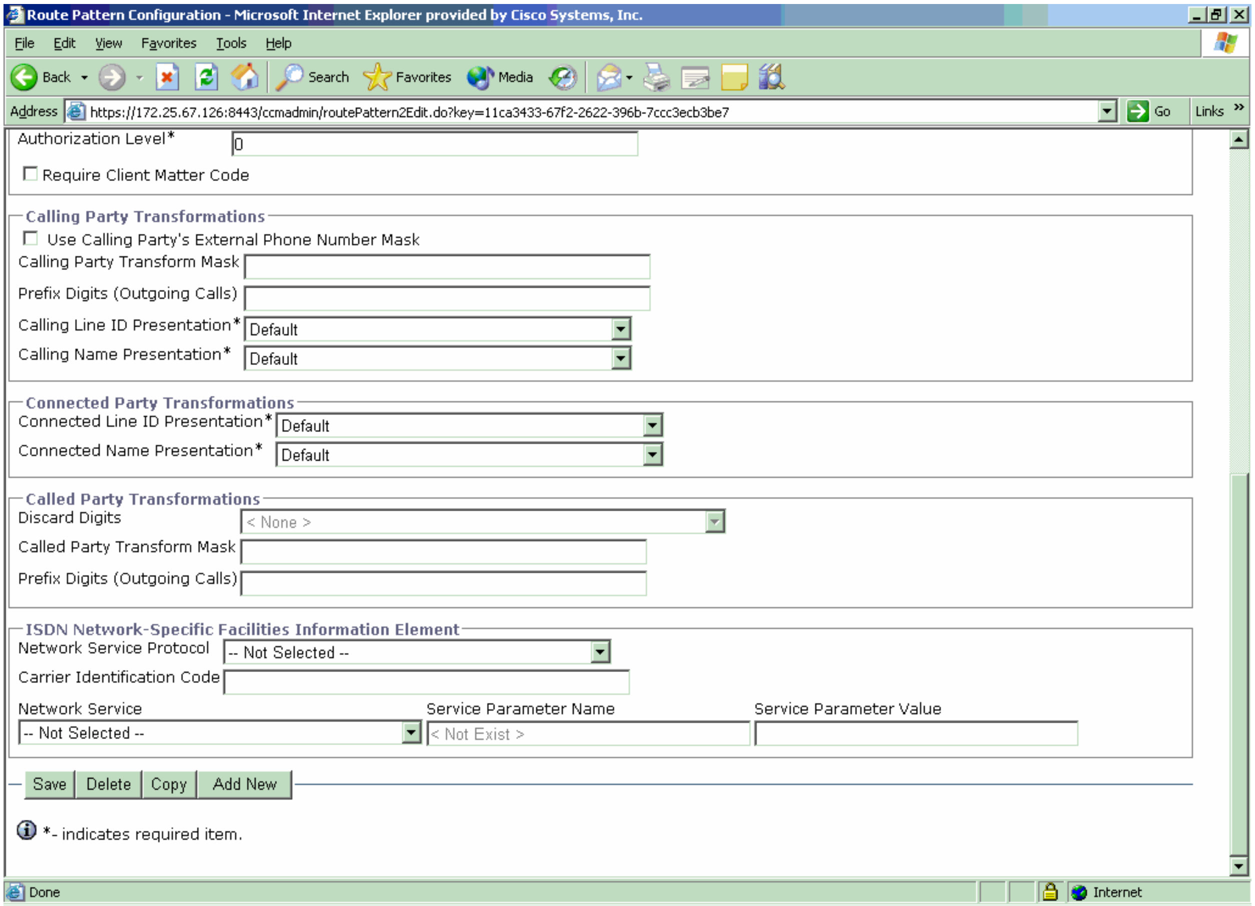
Save Delete Copy Reset Add New

*- indicates required item.

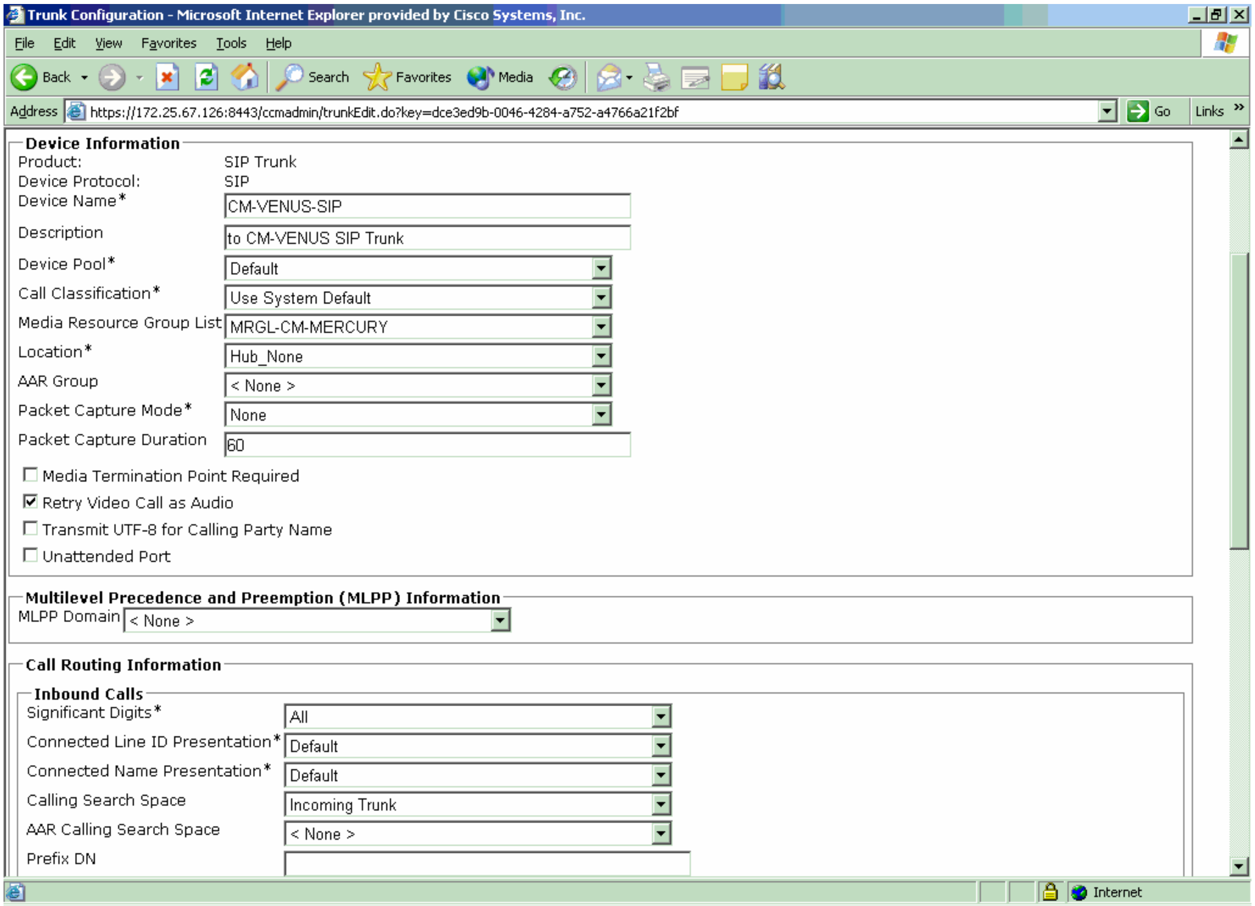
Done Internet

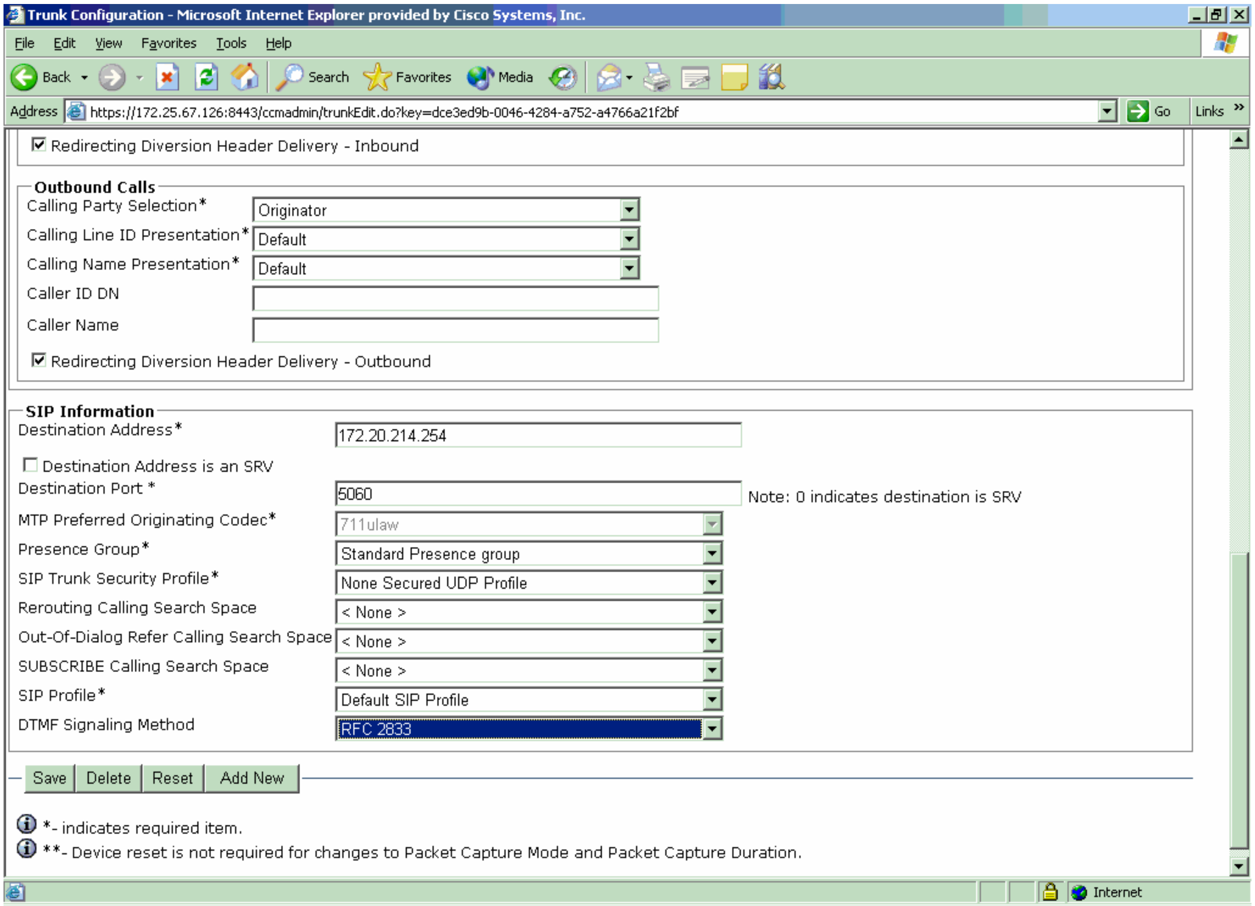
12.1.4 Route Pattern for Unity Access



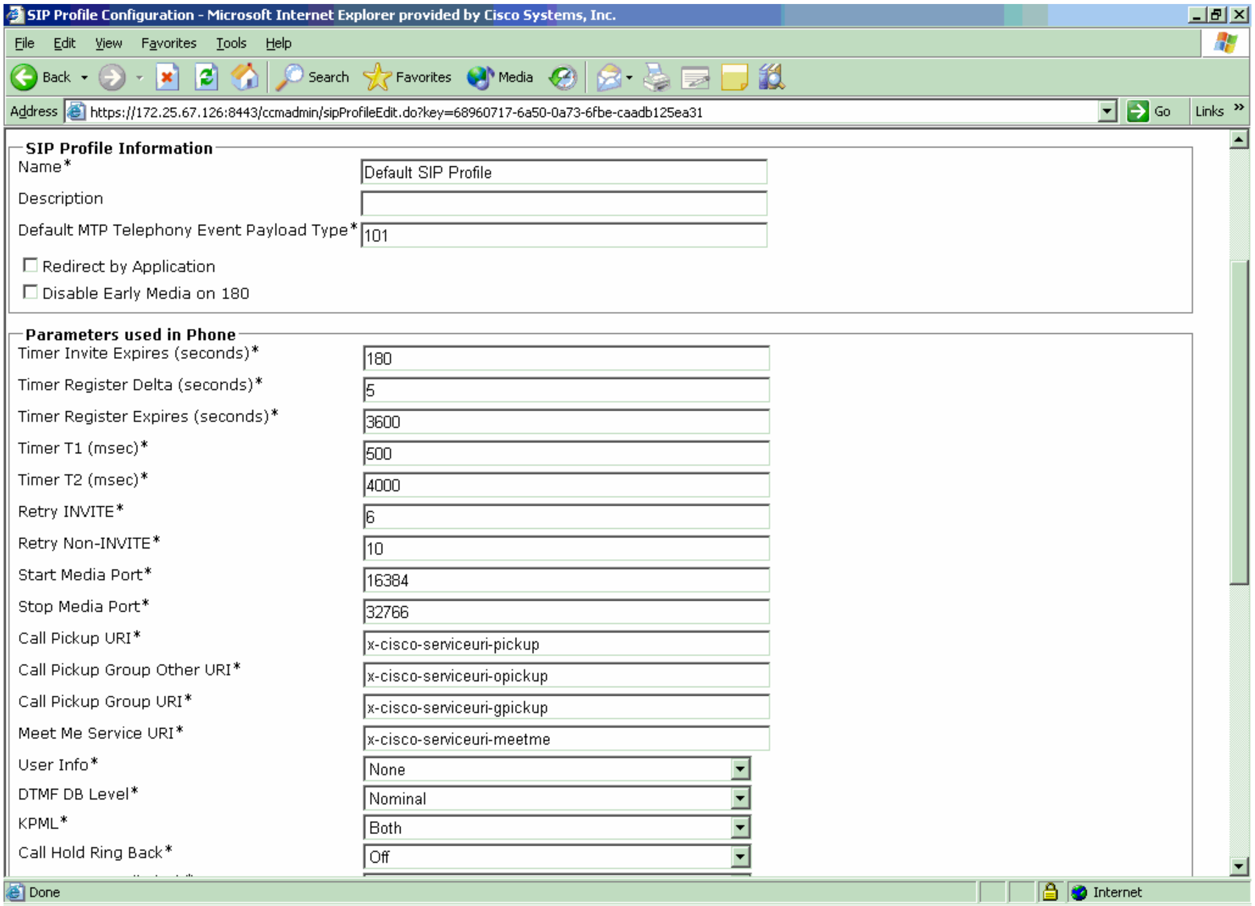

Route Pattern Configuration - Microsoft Internet Explorer provided by Cisco Systems, Inc.
File Edit View Favorites Tools Help
Back Search Favorites Media
Address https://172.25.67.126:8443/ccmadmin/routePattern2Edit.do?key=11ca3433-67f2-2622-396b-7ccc3ecb3be7
Go Links
Authorization Level* 0
Require Client Matter Code
Calling Party Transformations
Use Calling Party's External Phone Number Mask
Calling Party Transform Mask
Prefix Digits (Outgoing Calls)
Calling Line ID Presentation* Default
Calling Name Presentation* Default
Connected Party Transformations
Connected Line ID Presentation* Default
Connected Name Presentation* Default
Called Party Transformations
Discard Digits < None >
Called Party Transform Mask
Prefix Digits (Outgoing Calls)
ISDN Network-Specific Facilities Information Element
Network Service Protocol -- Not Selected --
Carrier Identification Code
Network Service
Service Parameter Name
Service Parameter Value
-- Not Selected --
< Not Exist >
Save Delete Copy Add New
*- indicates required item.
Done
Internet

12.1.5 SIP Trunk for Unity Access Configuration




Trunk Configuration - Microsoft Internet Explorer provided by Cisco Systems, Inc.
File Edit View Favorites Tools Help
Back Search Favorites Media
Address https://172.25.67.126:8443/ccmadmin/trunkEdit.do?key=dce3ed9b-0046-4284-a752-a4766a21f2bf
Go Links
Redirecting Diversion Header Delivery - Inbound
Outbound Calls
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* 172.20.214.254
Destination Address is an SRV
Destination Port * 5060 Note: 0 indicates destination is SRV
MTP Preferred Originating Codec* 711ulaw
Presence Group* Standard Presence group
SIP Trunk Security Profile* None Secured UDP Profile
Rerouting Calling Search Space < None >
Out-Of-Dialog Refer Calling Search Space < None >
SUBSCRIBE Calling Search Space < None >
SIP Profile* Default SIP Profile
DTMF Signaling Method RFC 2833
Save Delete Reset Add New
*- indicates required item.
**- Device reset is not required for changes to Packet Capture Mode and Packet Capture Duration.
Internet

12.1.6 Default SIP Profile Configuration


SIP Profile Configuration - Microsoft Internet Explorer provided by Cisco Systems, Inc.

File Edit View Favorites Tools Help

Address https://172.25.67.126:8443/ccmadmin/sipProfileEdit.do?key=68960717-6a50-0a73-6fbe-caadb125ea31

SIP Profile Information

Name*	Default SIP Profile
Description	
Default MTP Telephony Event Payload Type*	101

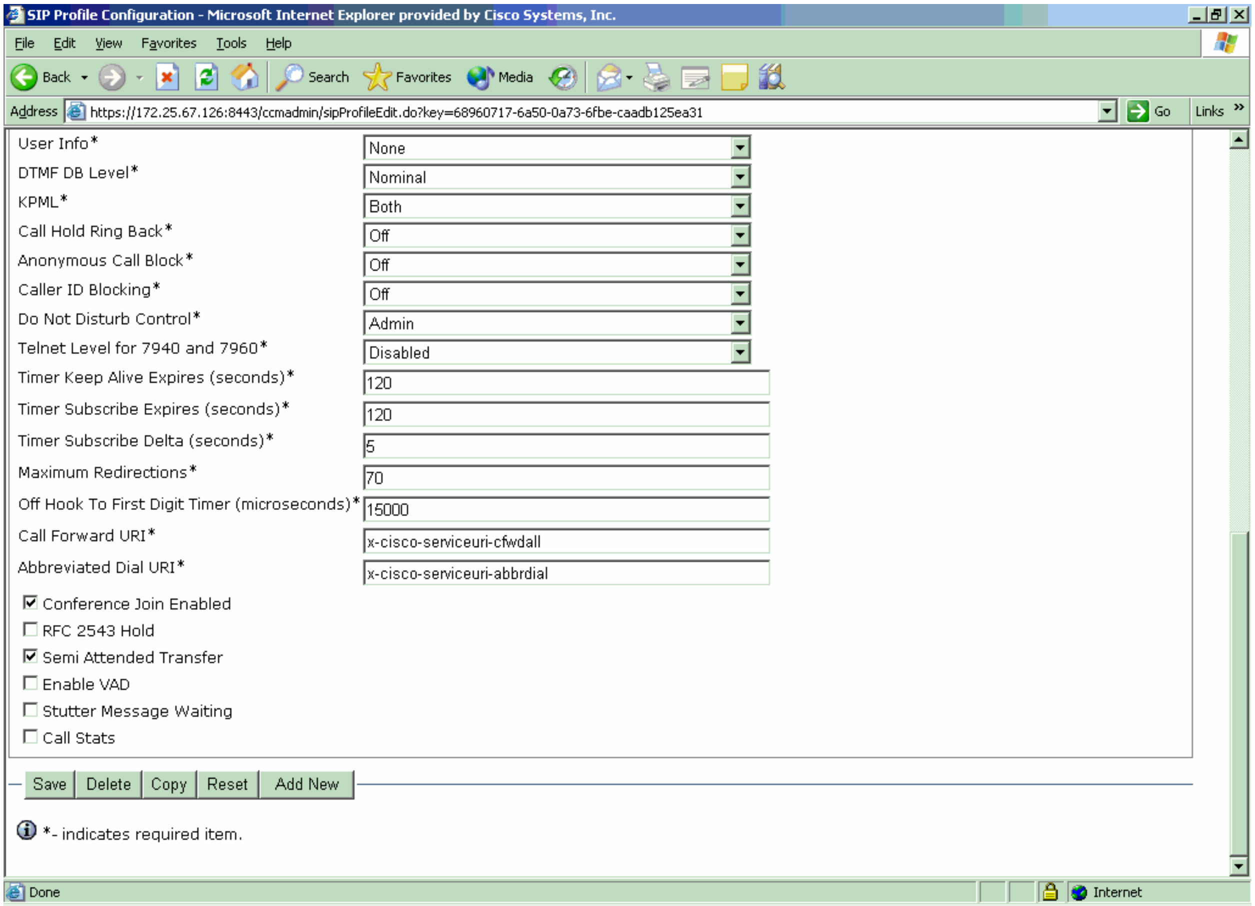
☐ Redirect by Application

☐ Disable Early Media on 180

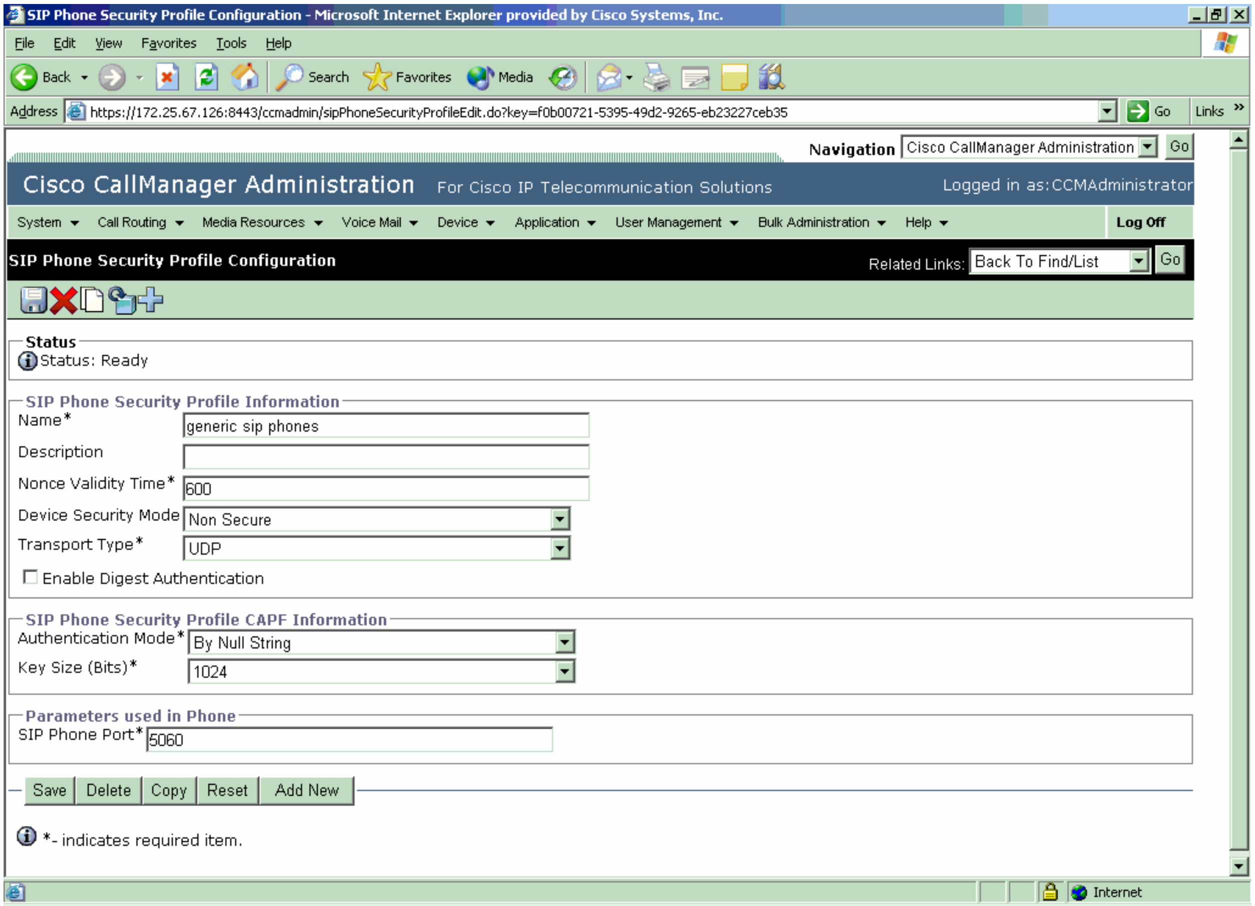
Parameters used in Phone

Timer Invite Expires (seconds)*	180
Timer Register Delta (seconds)*	5
Timer Register Expires (seconds)*	3600
Timer T1 (msec)*	500
Timer T2 (msec)*	4000
Retry INVITE*	6
Retry Non-INVITE*	10
Start Media Port*	16384
Stop Media Port*	32766
Call Pickup URI*	x-cisco-serviceuri-pickup
Call Pickup Group Other URI*	x-cisco-serviceuri-opickup
Call Pickup Group URI*	x-cisco-serviceuri-gpickup
Meet Me Service URI*	x-cisco-serviceuri-meetme
User Info*	None
DTMF DB Level*	Nominal
KPML*	Both
Call Hold Ring Back*	Off

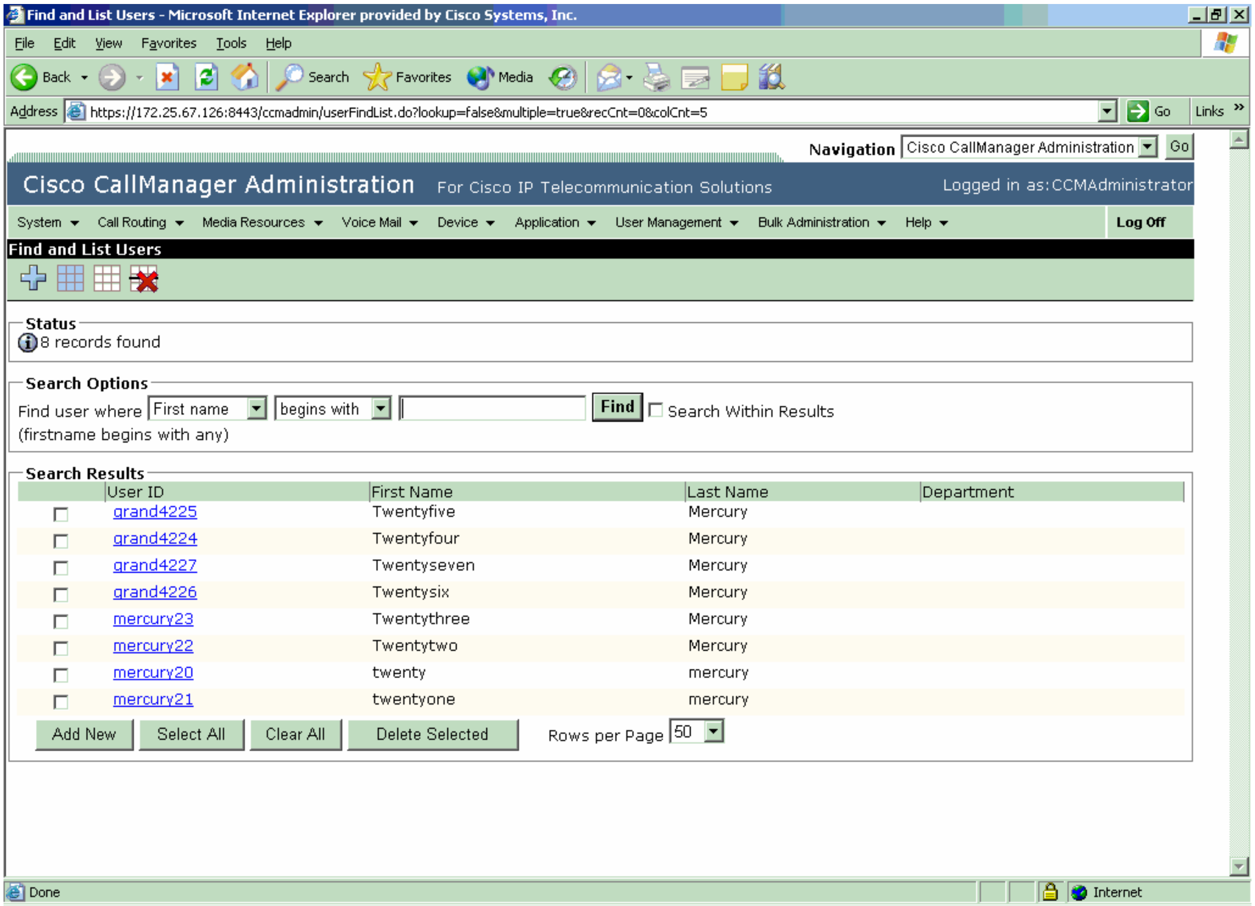
Done | Internet


SIP Profile Configuration - Microsoft Internet Explorer provided by Cisco Systems, Inc.
File Edit View Favorites Tools Help
Back Search Favorites Media
Address https://172.25.67.126:8443/ccmadmin/sipProfileEdit.do?key=68960717-6a50-0a73-6fbe-caadb125ea31
Go Links
User Info* None
DTMF DB Level* Nominal
KPML* Both
Call Hold Ring Back* Off
Anonymous Call Block* Off
Caller ID Blocking* Off
Do Not Disturb Control* Admin
Telnet Level for 7940 and 7960* Disabled
Timer Keep Alive Expires (seconds)* 120
Timer Subscribe Expires (seconds)* 120
Timer Subscribe Delta (seconds)* 5
Maximum Redirections* 70
Off Hook To First Digit Timer (microseconds)* 15000
Call Forward URI* x-cisco-serviceuri-cfwdall
Abbreviated Dial URI* x-cisco-serviceuri-abbrdial
Conference Join Enabled
RFC 2543 Hold
Semi Attended Transfer
Enable VAD
Stutter Message Waiting
Call Stats
Save Delete Copy Reset Add New
*- indicates required item.
Done
Internet

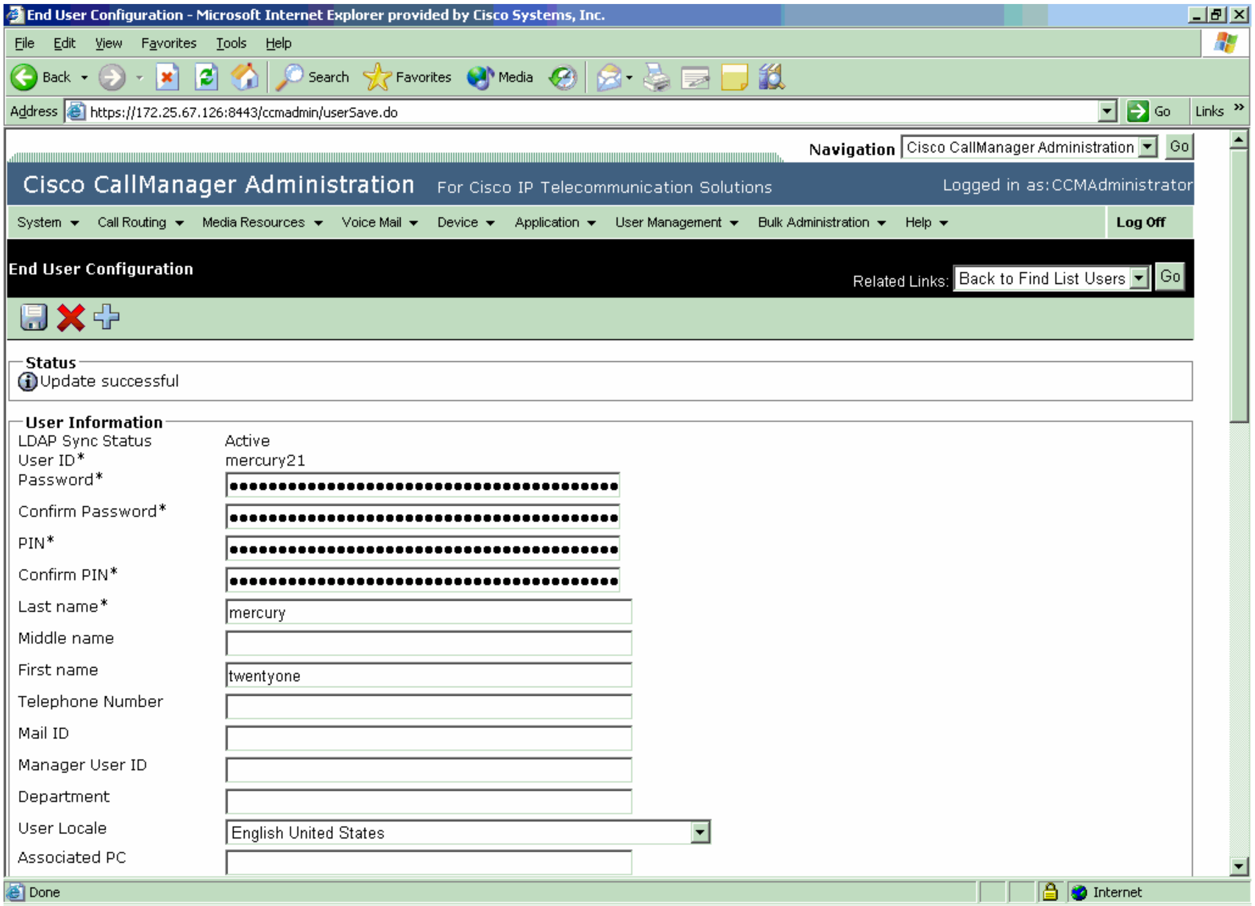
12.1.7 Generic SIP Phone Security Profile Configuration

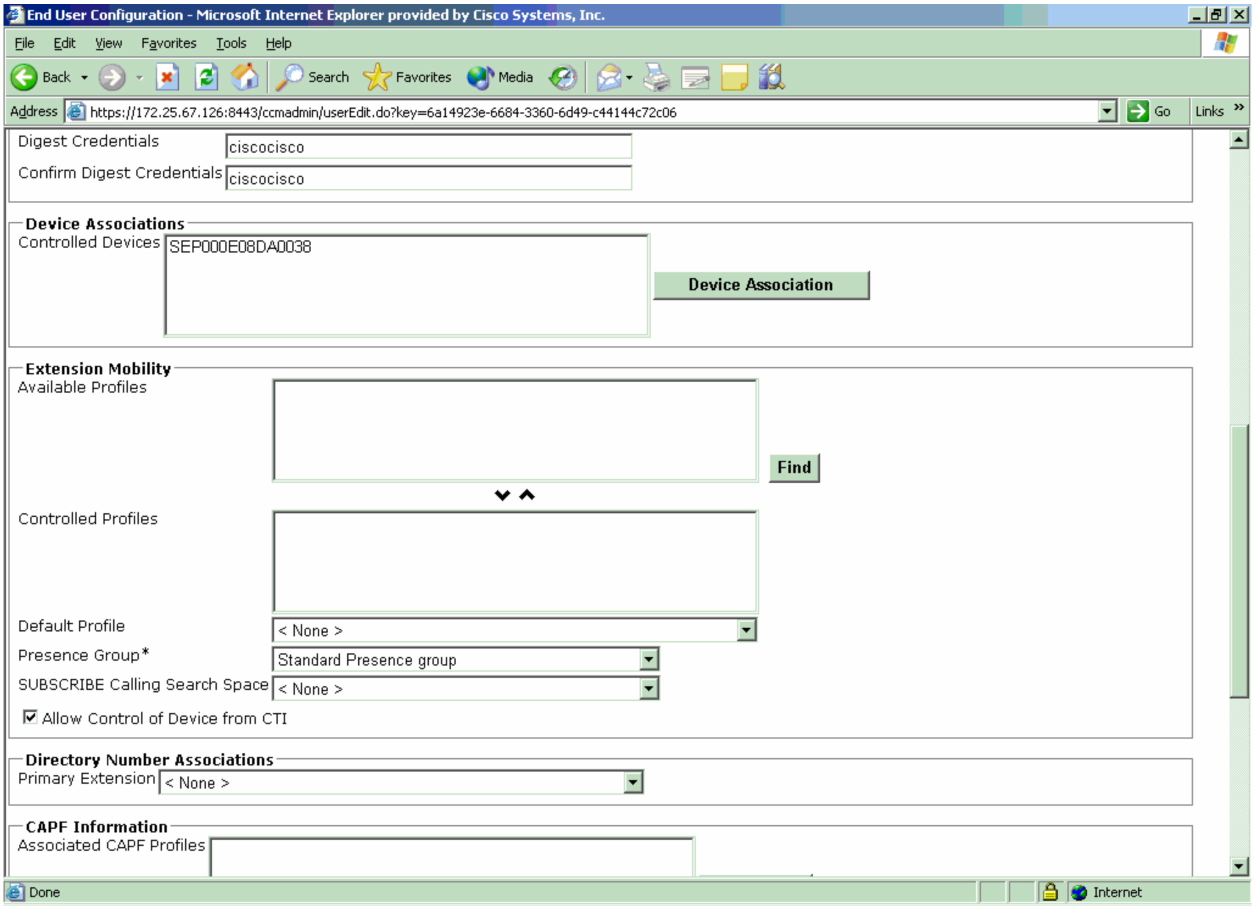


12.1.8 User ID List for 3rd Party Phones

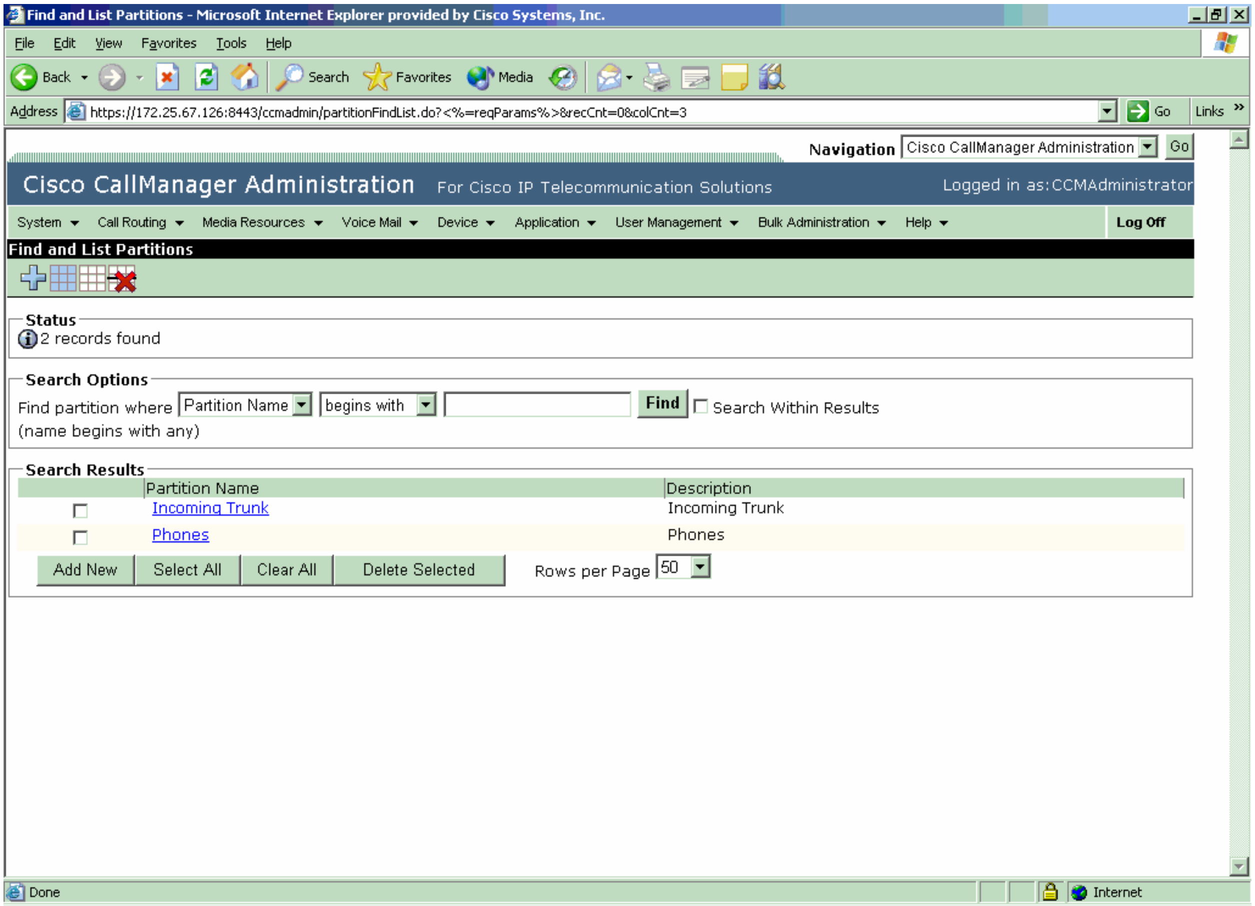


12.1.9 End User Configuration for 3rd Party Phone

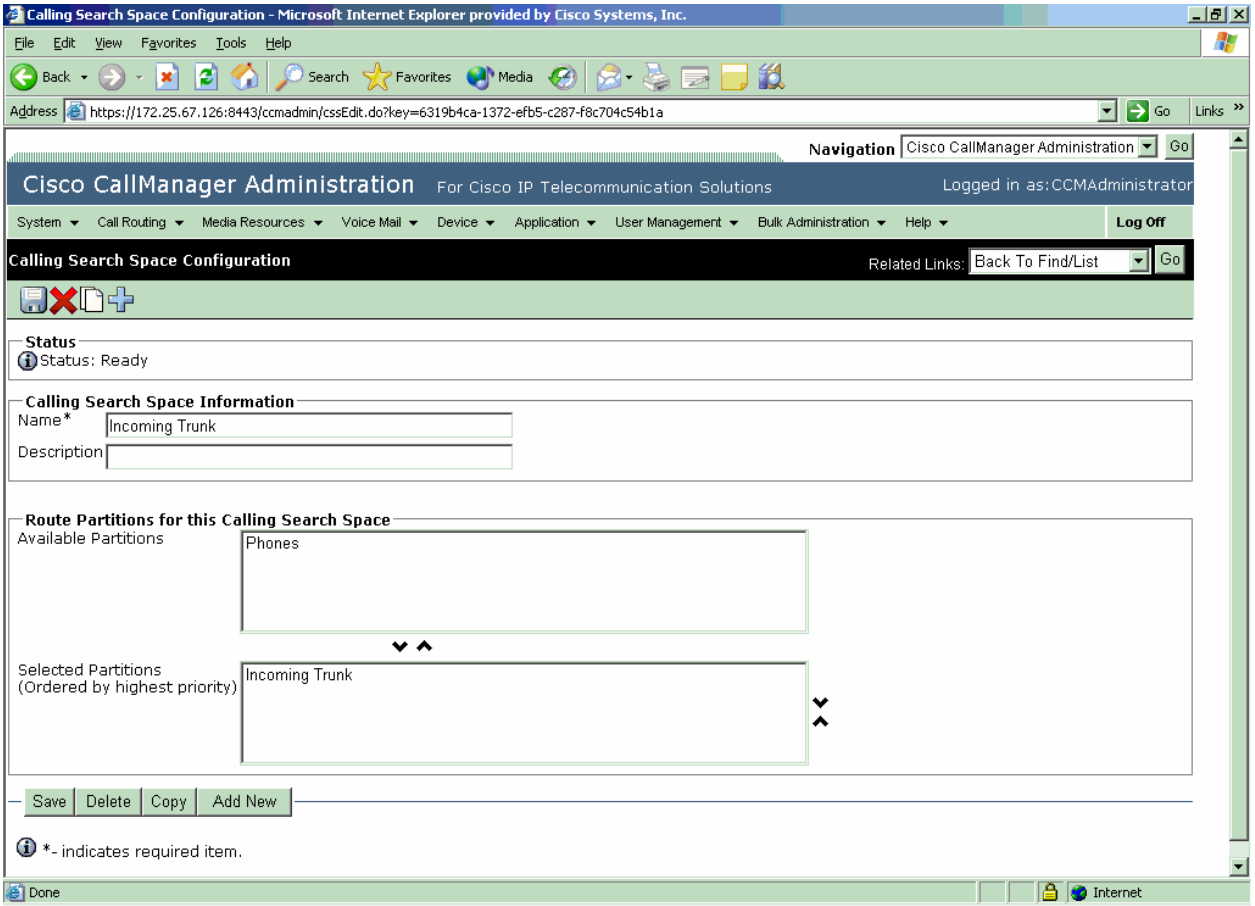



End User Configuration - Microsoft Internet Explorer provided by Cisco Systems, Inc.
File Edit View Favorites Tools Help
Back Search Favorites Media
Address https://172.25.67.126:8443/ccmadmin/userEdit.do?key=6a14923e-6684-3360-6d49-c44144c72c06
Go Links
Digest Credentials ciscocisco
Confirm Digest Credentials ciscocisco
Device Associations
Controlled Devices SEP000E08DA0038
Device Association
Extension Mobility
Available Profiles
Find
Controlled Profiles
Default Profile < None >
Presence Group* Standard Presence group
SUBSCRIBE Calling Search Space < None >
Allow Control of Device from CTI
Directory Number Associations
Primary Extension < None >
CAPF Information
Associated CAPF Profiles
Done
Internet

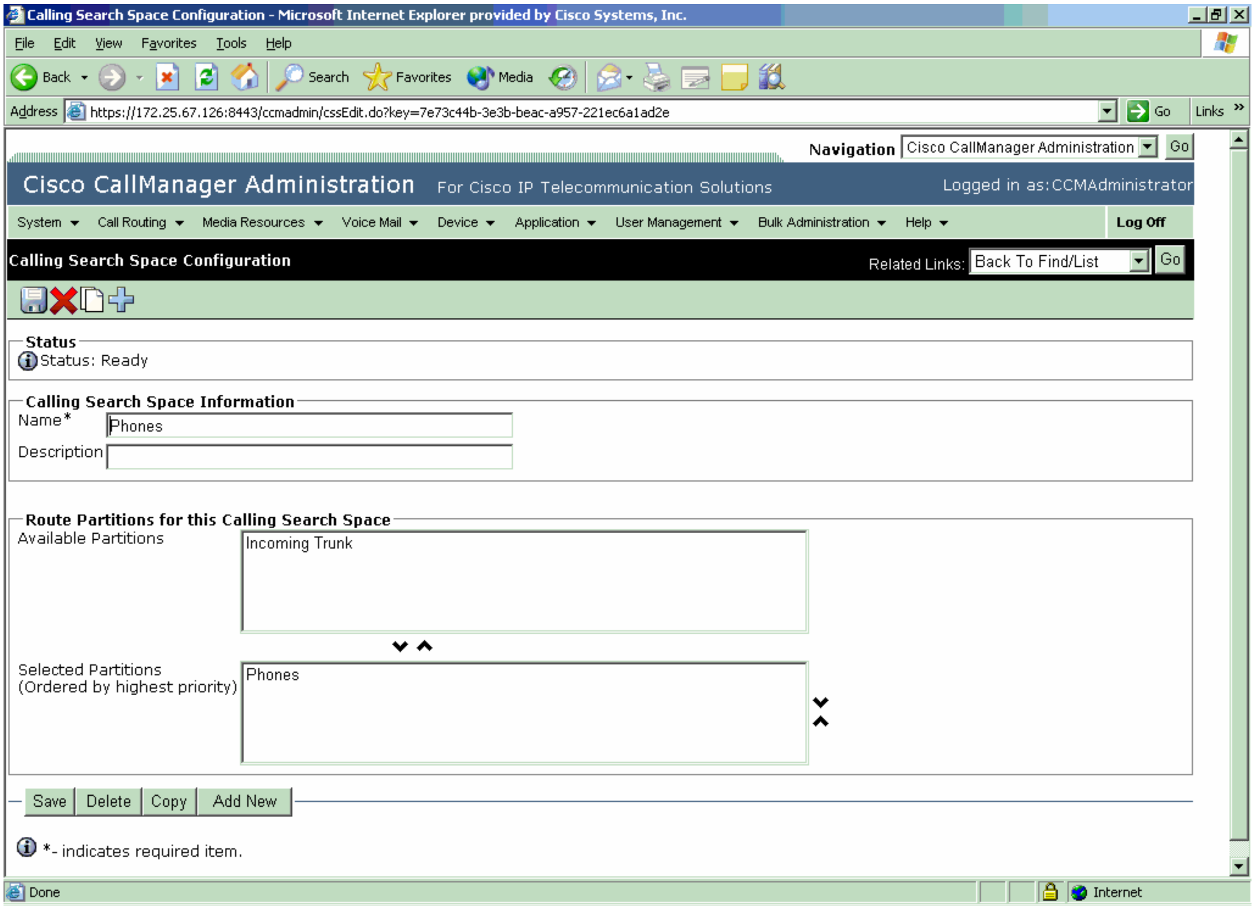
12.1.10Partitions Configuration



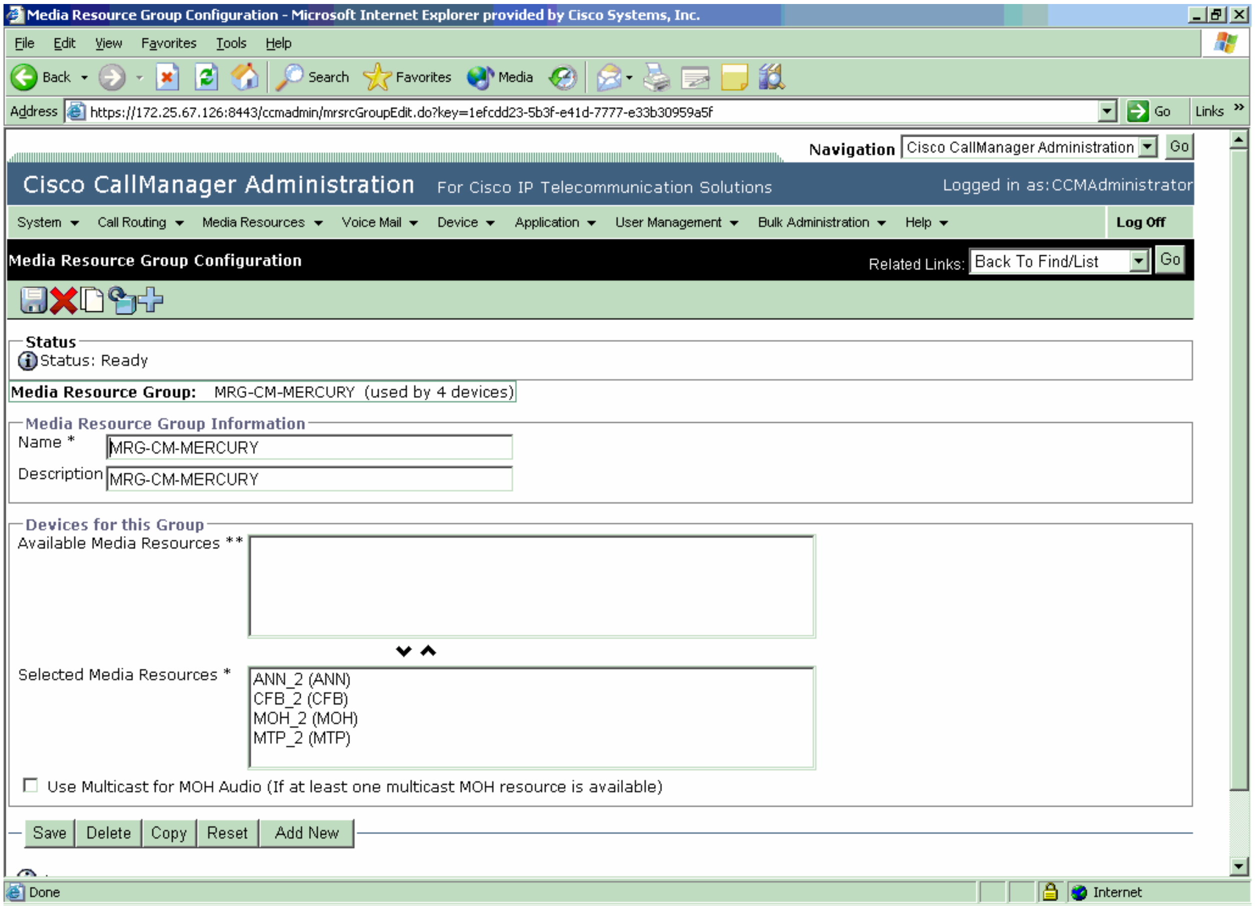
12.1.11 Incoming Trunk CSS Configuration



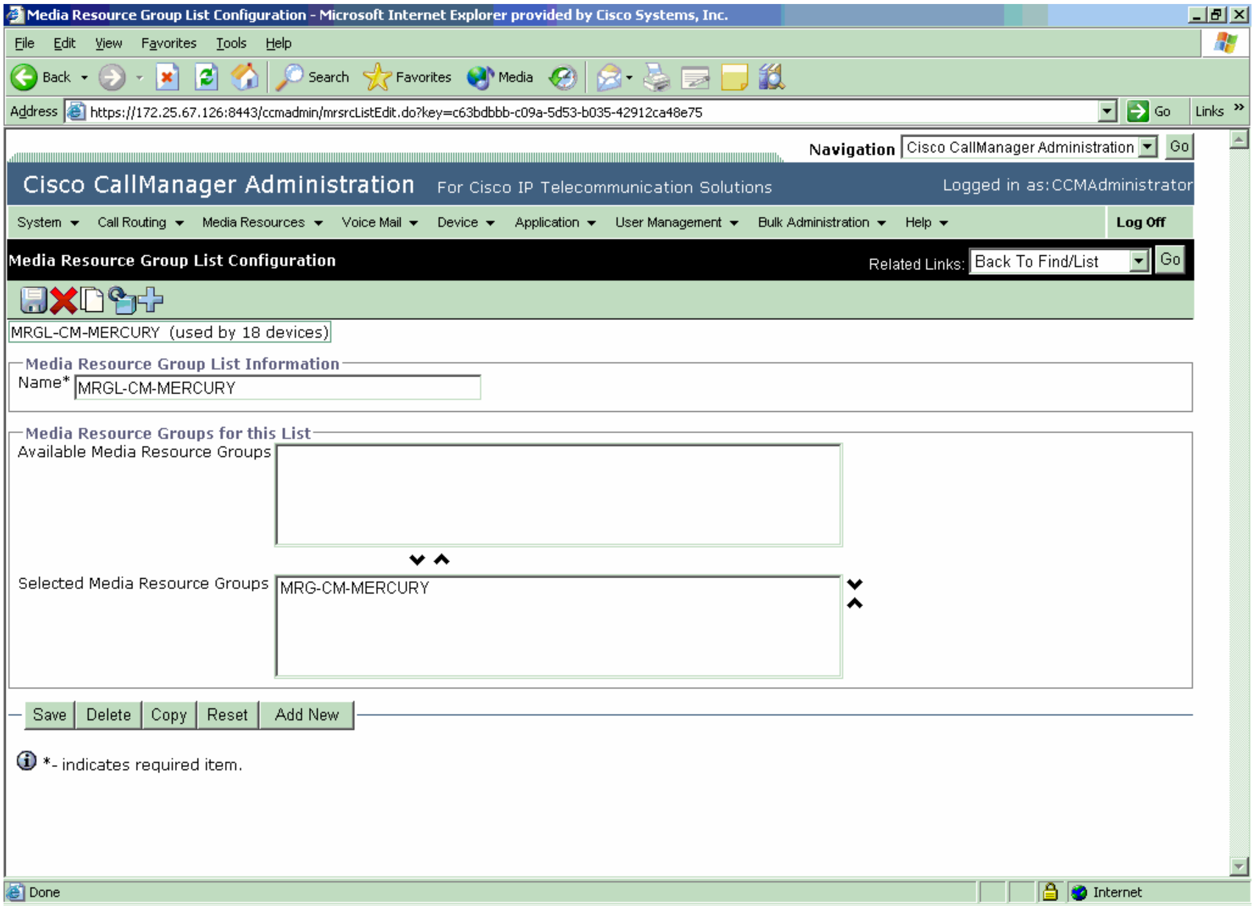
12.1.12 Phones CSS Configuration



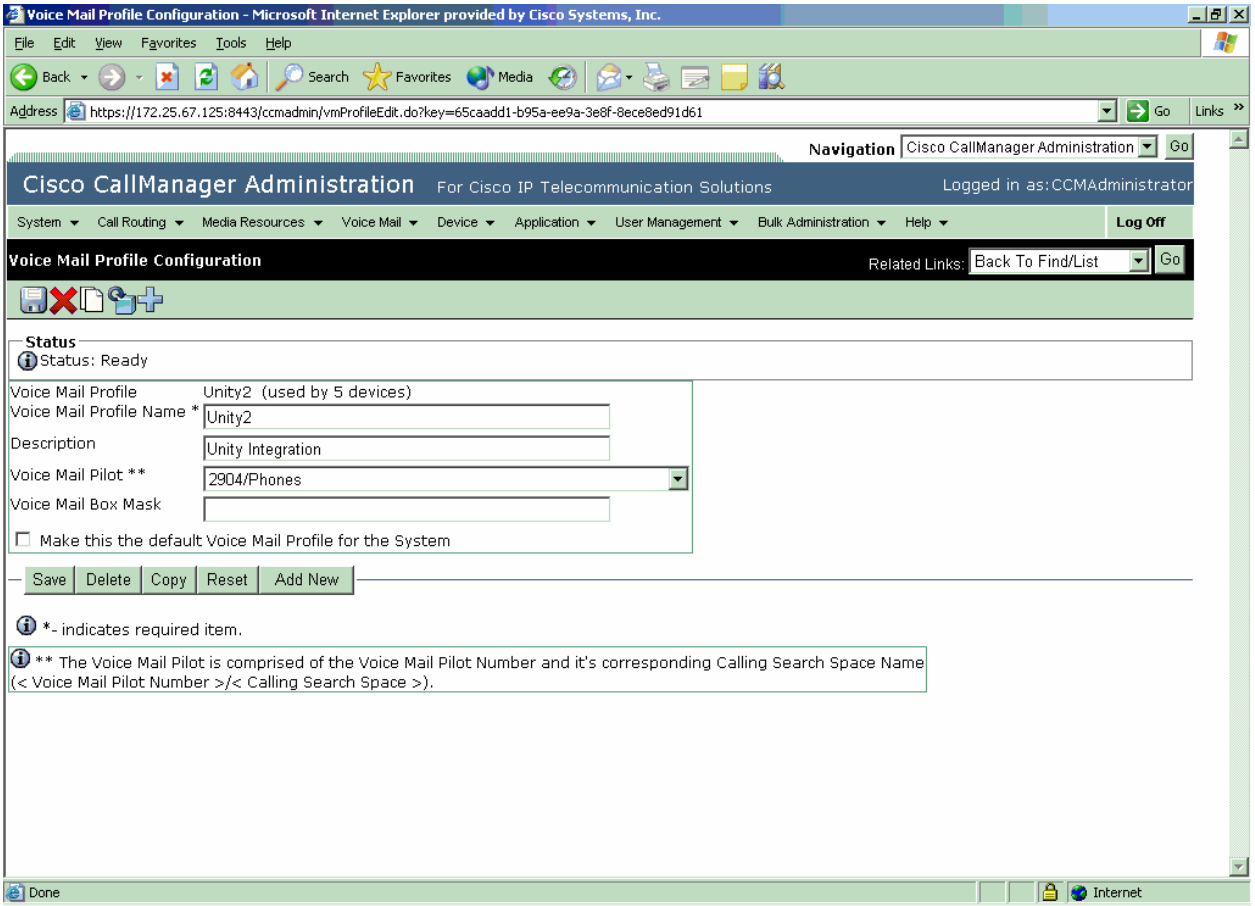
12.1.13 Media Resource Group



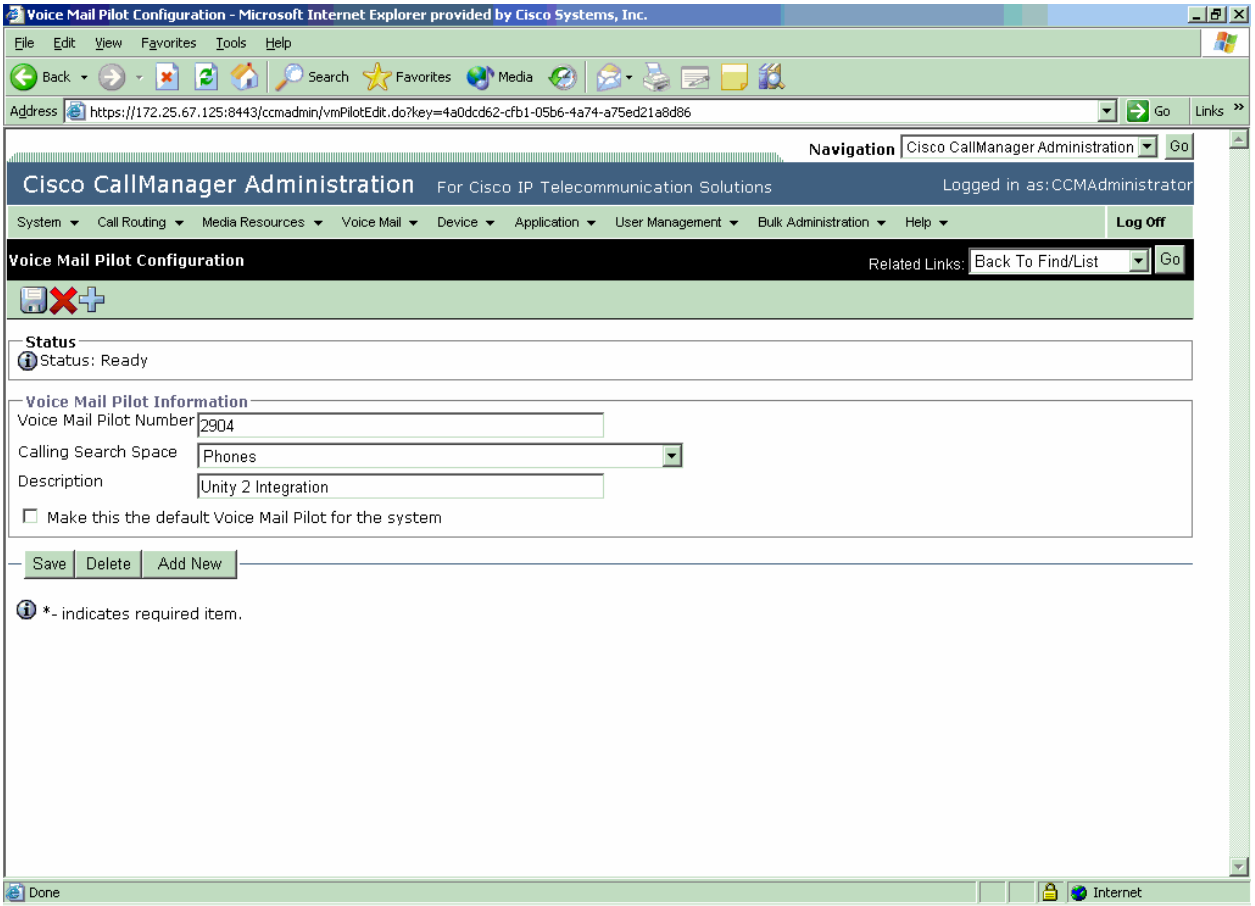
12.1.14 Media Resource Group List



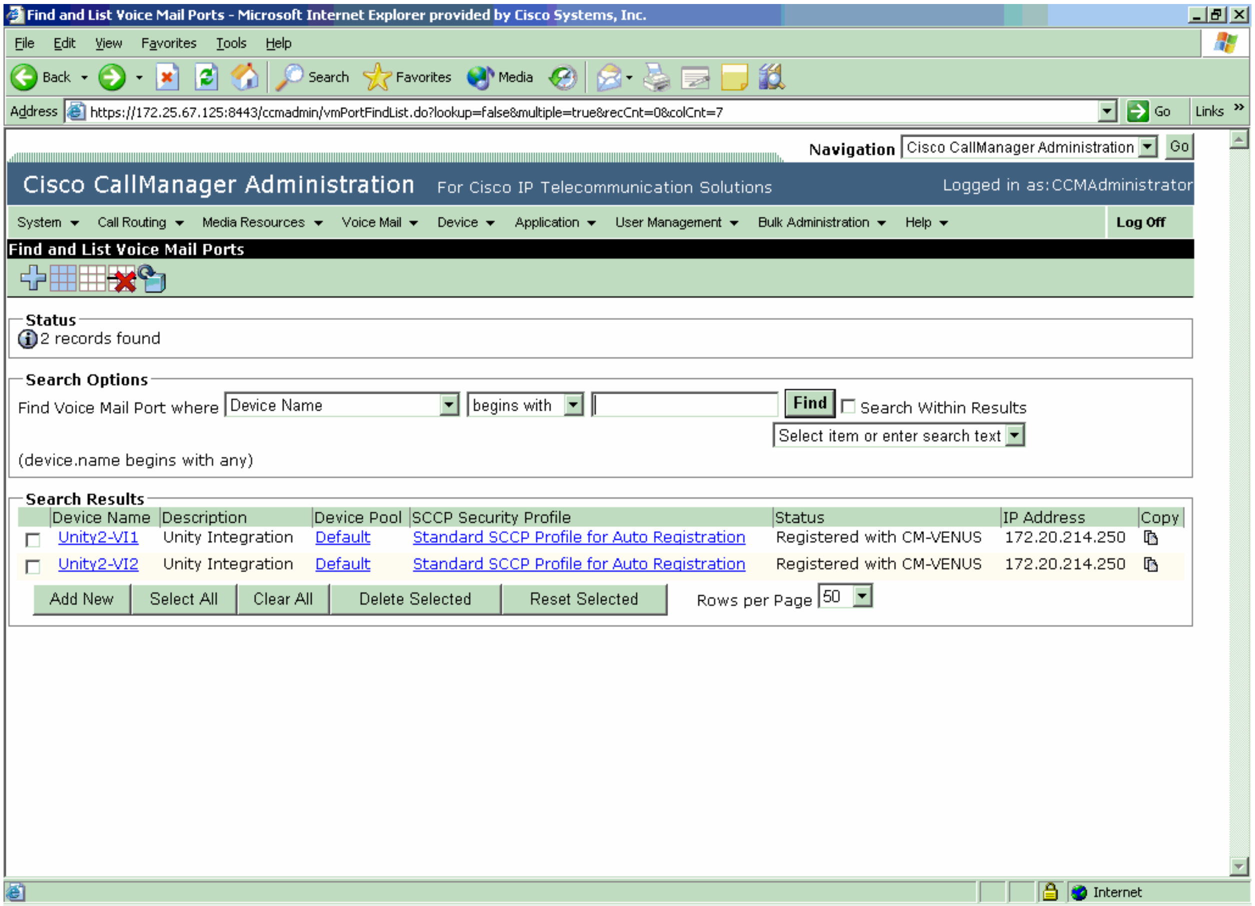
12.1.15 Voicemail Profile Configuration

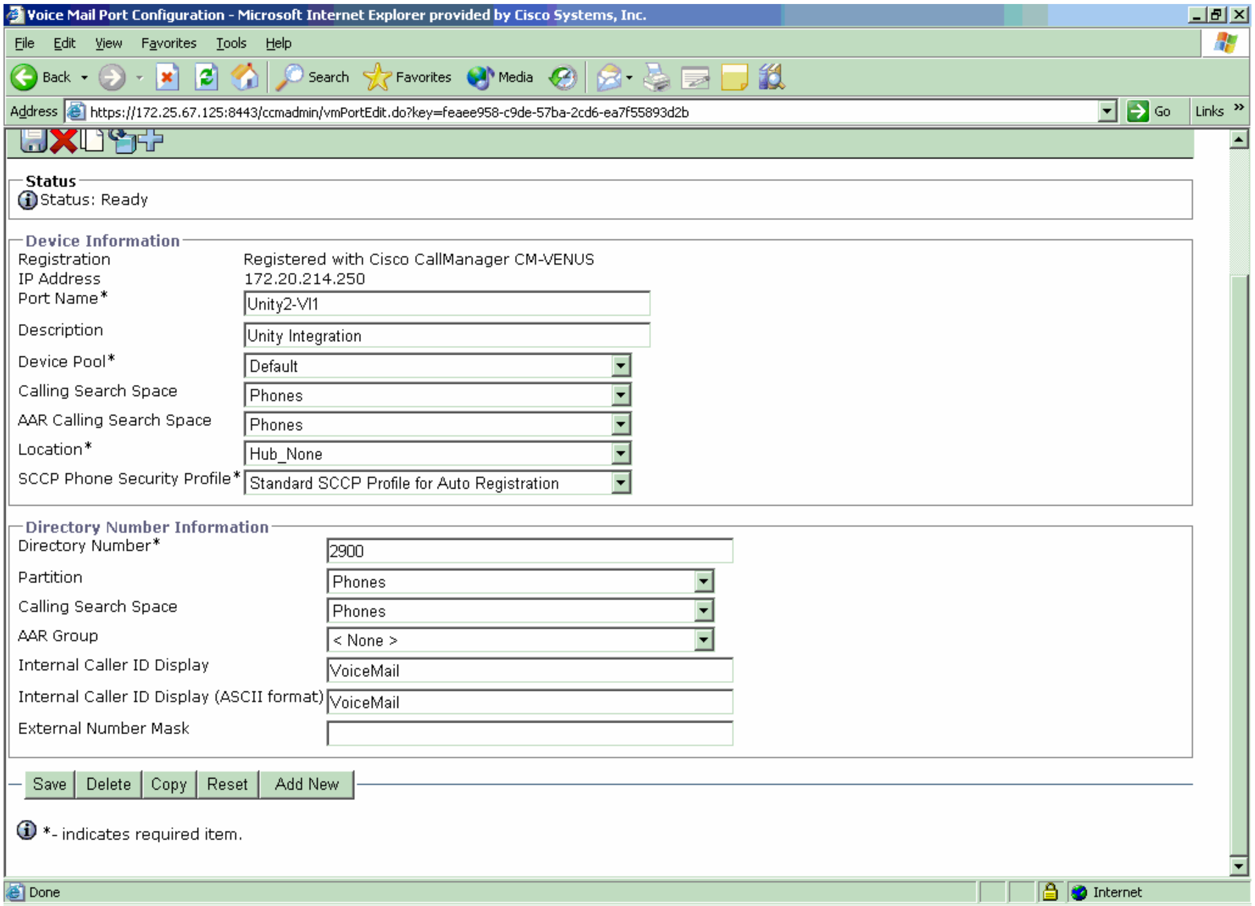


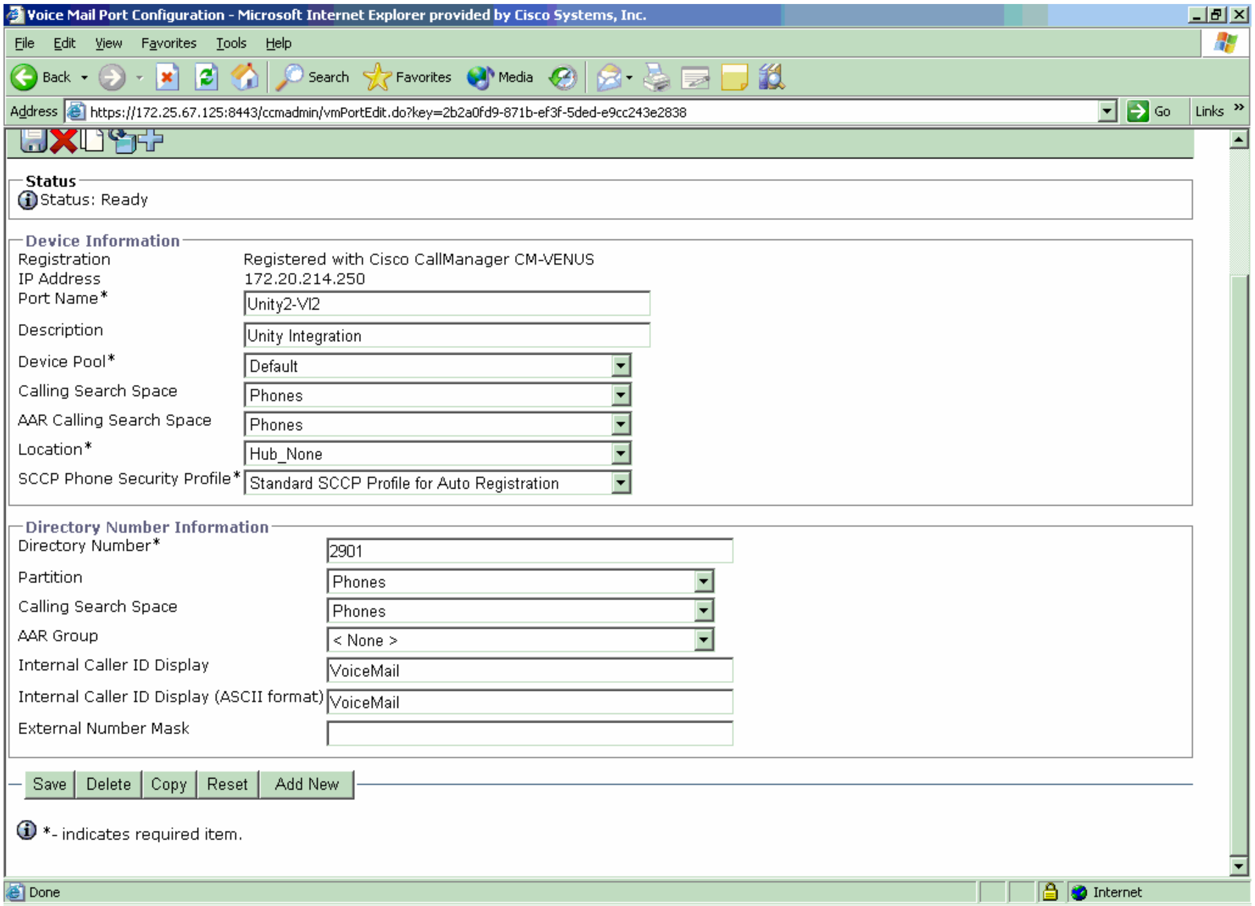
12.1.16 Voicemail Pilot



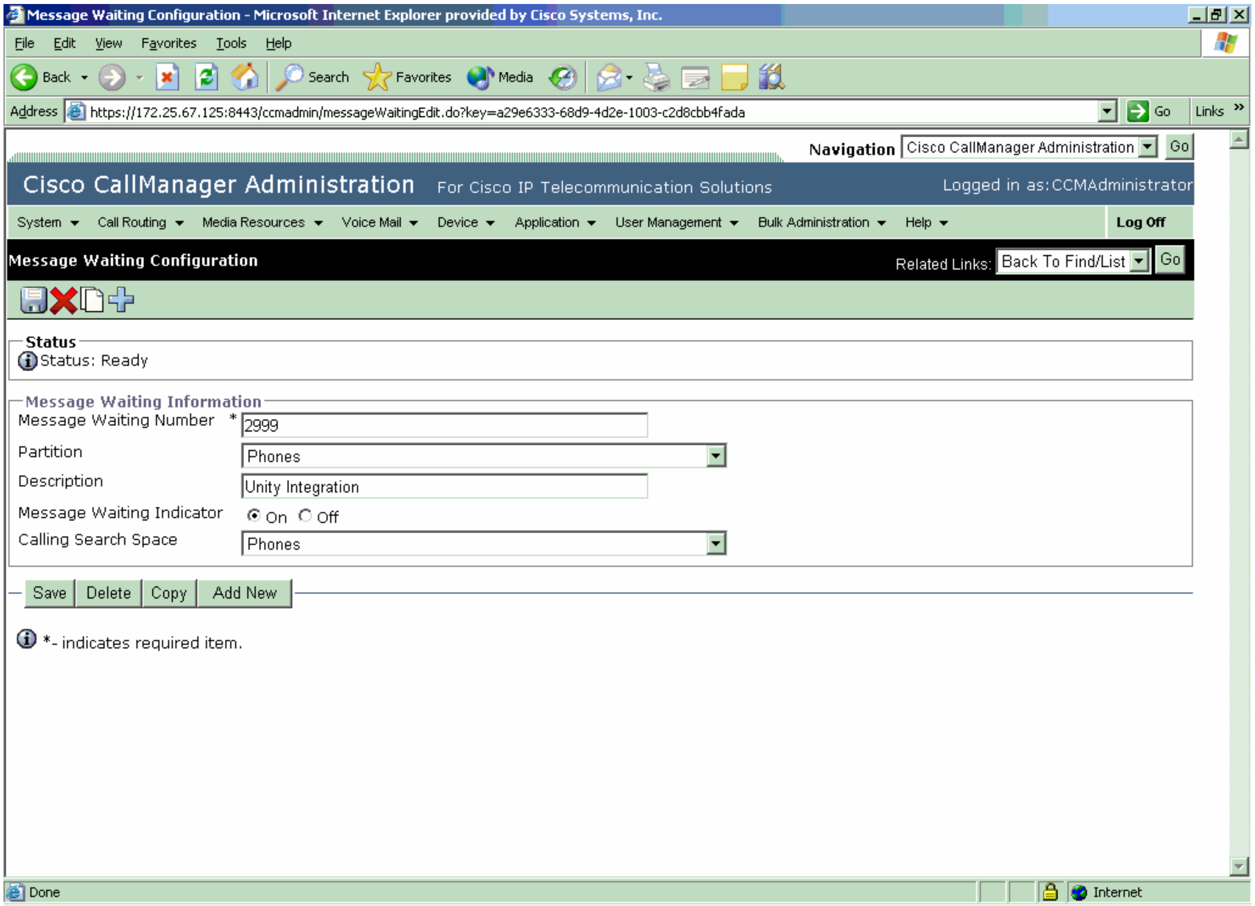
12.1.17 Voicemail Ports

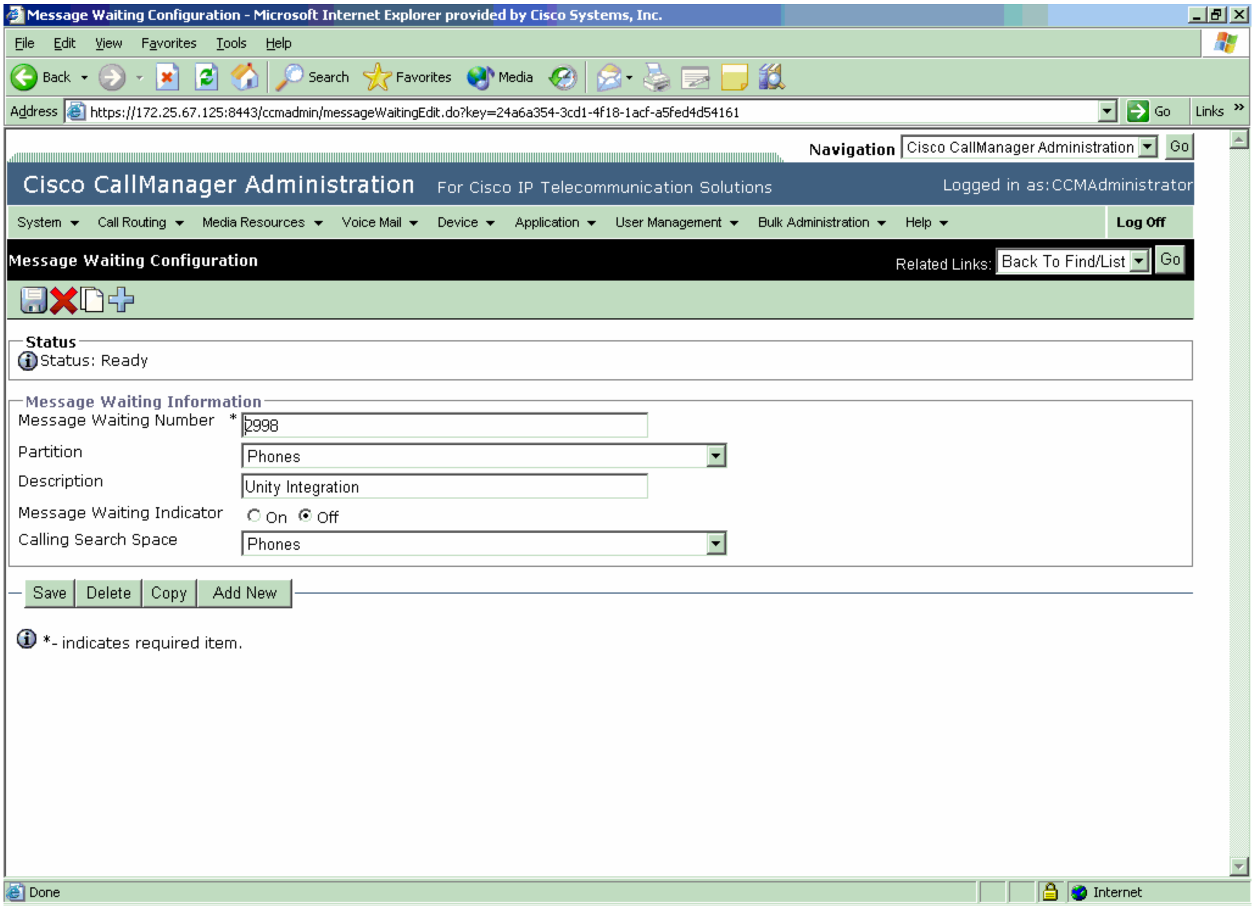



Voice Mail Port Configuration - Microsoft Internet Explorer provided by Cisco Systems, Inc.
File Edit View Favorites Tools Help
Back Search Favorites Media
Address https://172.25.67.125:8443/ccmadmin/vmPortEdit.do?key=feaee958-c9de-57ba-2cd6-ea7f55893d2b
Go Links
Status
Status: Ready
Device Information
Registration
Registered with Cisco CallManager CM-VENUS
IP Address
172.20.214.250
Port Name*
Unity2-VI1
Description
Unity Integration
Device Pool*
Default
Calling Search Space
Phones
AAR Calling Search Space
Phones
Location*
Hub_None
SCCP Phone Security Profile*
Standard SCCP Profile for Auto Registration
Directory Number Information
Directory Number*
2900
Partition
Phones
Calling Search Space
Phones
AAR Group
< None >
Internal Caller ID Display
VoiceMail
Internal Caller ID Display (ASCII format)
VoiceMail
External Number Mask
Save Delete Copy Reset Add New
*- indicates required item.
Done
Internet

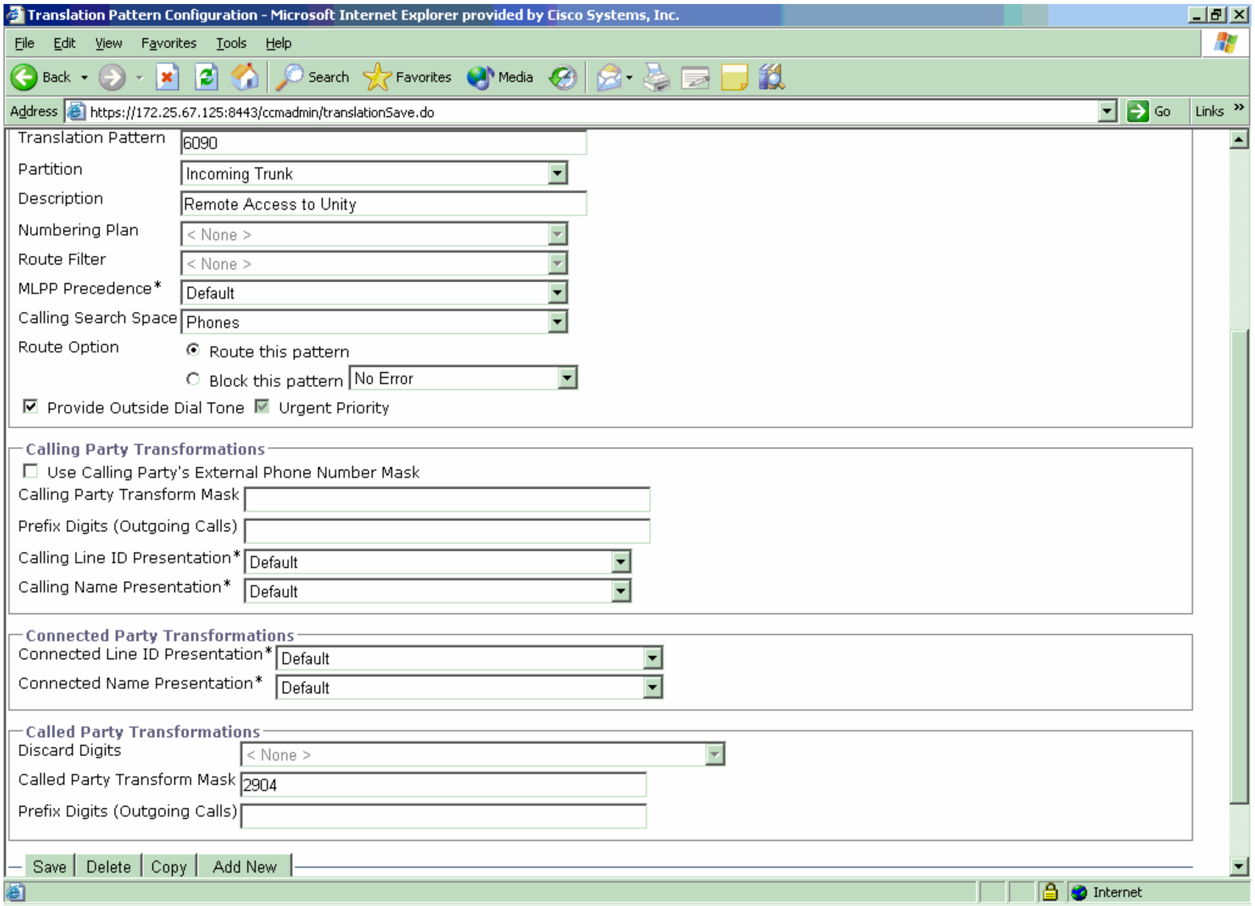

Voice Mail Port Configuration - Microsoft Internet Explorer provided by Cisco Systems, Inc.
File Edit View Favorites Tools Help
Back Search Favorites Media
Address https://172.25.67.125:8443/ccmadmin/vmPortEdit.do?key=2b2a0fd9-871b-ef3f-5ded-e9cc243e2838
Go Links
Status
Status: Ready
Device Information
Registration Registered with Cisco CallManager CM-VENUS
IP Address 172.20.214.250
Port Name* Unity2-VI2
Description Unity Integration
Device Pool* Default
Calling Search Space Phones
AAR Calling Search Space Phones
Location* Hub_None
SCCP Phone Security Profile* Standard SCCP Profile for Auto Registration
Directory Number Information
Directory Number* 2901
Partition Phones
Calling Search Space Phones
AAR Group < None >
Internal Caller ID Display VoiceMail
Internal Caller ID Display (ASCII format) VoiceMail
External Number Mask
Save Delete Copy Reset Add New
*- indicates required item.
Done
Internet

12.1.18 MWI Configuration




Message Waiting Configuration - Microsoft Internet Explorer provided by Cisco Systems, Inc.
File Edit View Favorites Tools Help
Back Search Favorites Media
Address https://172.25.67.125:8443/ccmadmin/messageWaitingEdit.do?key=24a6a354-3cd1-4f18-1acf-a5fed4d54161
Go Links
Navigation Cisco CallManager Administration Go
Cisco CallManager Administration
For Cisco IP Telecommunication Solutions
Logged in as:CCMAdministrator
System Call Routing Media Resources Voice Mail Device Application User Management Bulk Administration Help
Log Off
Message Waiting Configuration
Related Links: Back To Find/List Go
Status
Status: Ready
Message Waiting Information
Message Waiting Number * 2998
Partition Phones
Description Unity Integration
Message Waiting Indicator On Off
Calling Search Space Phones
Save Delete Copy Add New
*- indicates required item.
Done
Internet

12.1.19 Translation Pattern for Voicemail Access


Translation Pattern Configuration - Microsoft Internet Explorer provided by Cisco Systems, Inc.

Address: https://172.25.67.125:8443/ccmadmin/translationSave.do

Translation Pattern: 6090
Partition: Incoming Trunk
Description: Remote Access to Unity
Numbering Plan: < None >
Route Filter: < None >
MLPP Precedence*: Default
Calling Search Space: Phones
Route Option: (•) Route this pattern ( ) Block this pattern No Error
[x] Provide Outside Dial Tone [x] Urgent Priority

Calling Party Transformations
[ ] Use Calling Party's External Phone Number Mask
Calling Party Transform Mask:
Prefix Digits (Outgoing Calls):
Calling Line ID Presentation*: Default
Calling Name Presentation*: Default

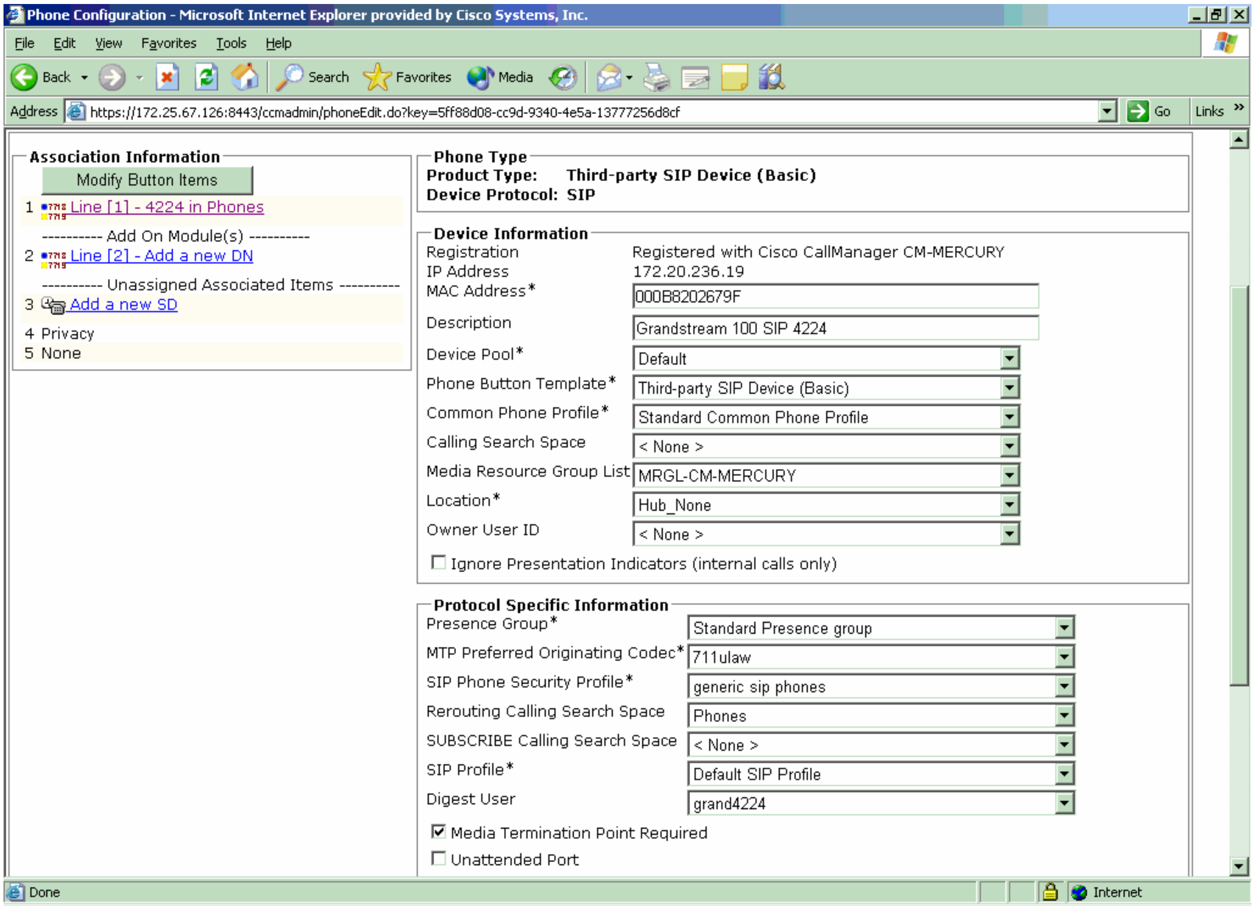
Connected Party Transformations
Connected Line ID Presentation*: Default
Connected Name Presentation*: Default

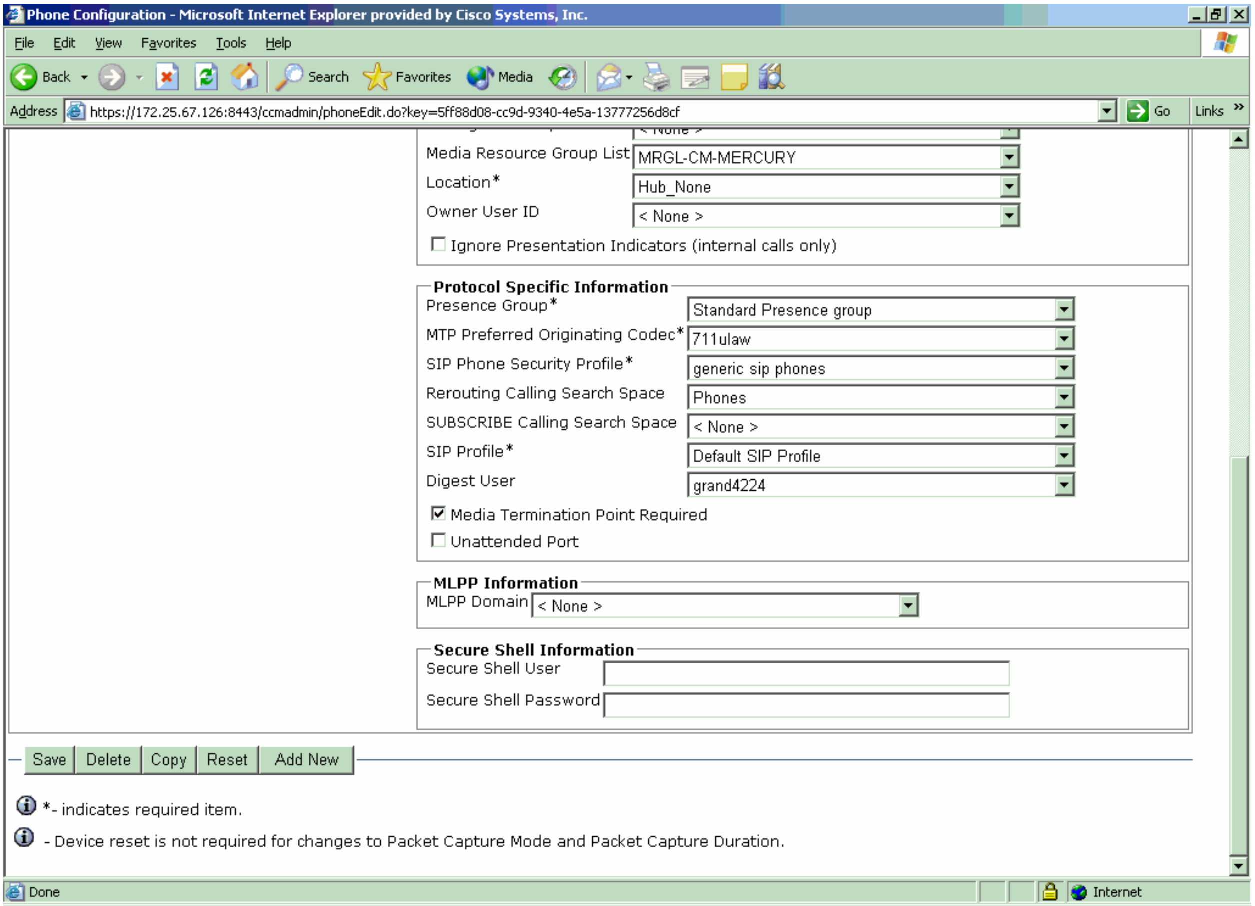
Called Party Transformations
Discard Digits: < None >
Called Party Transform Mask: 2904
Prefix Digits (Outgoing Calls):

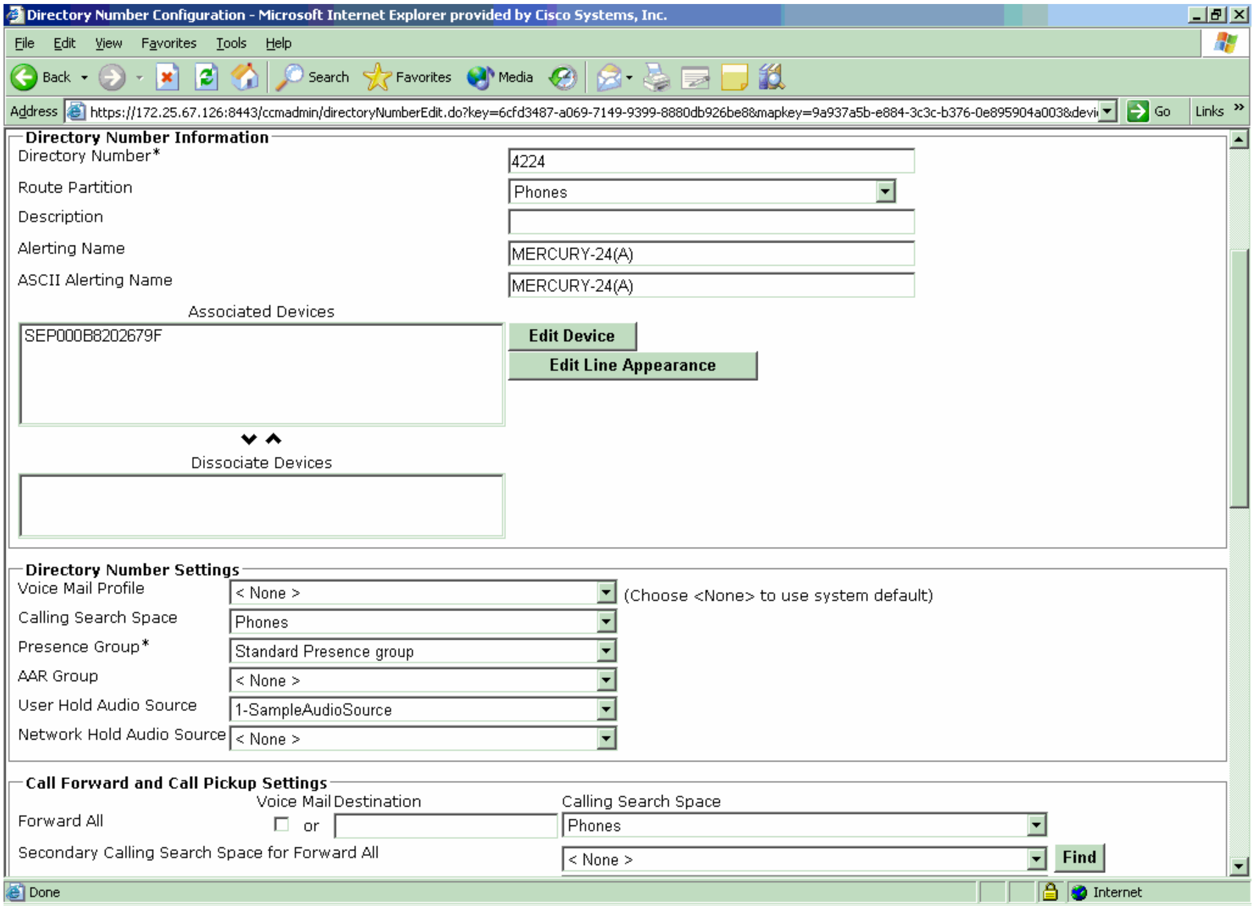
Save | Delete | Copy | Add New

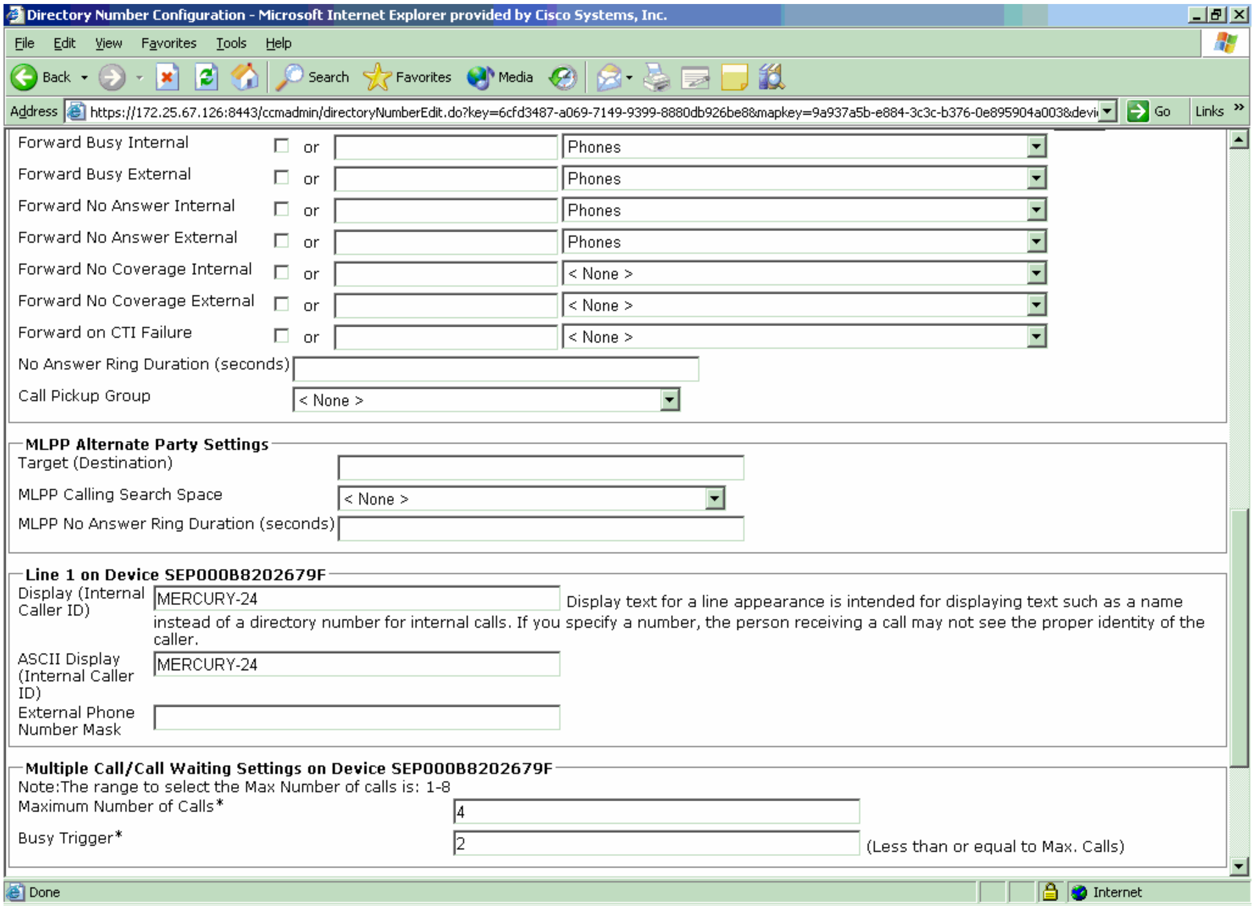
12.2 3rd party SIP Phones CCM configuration

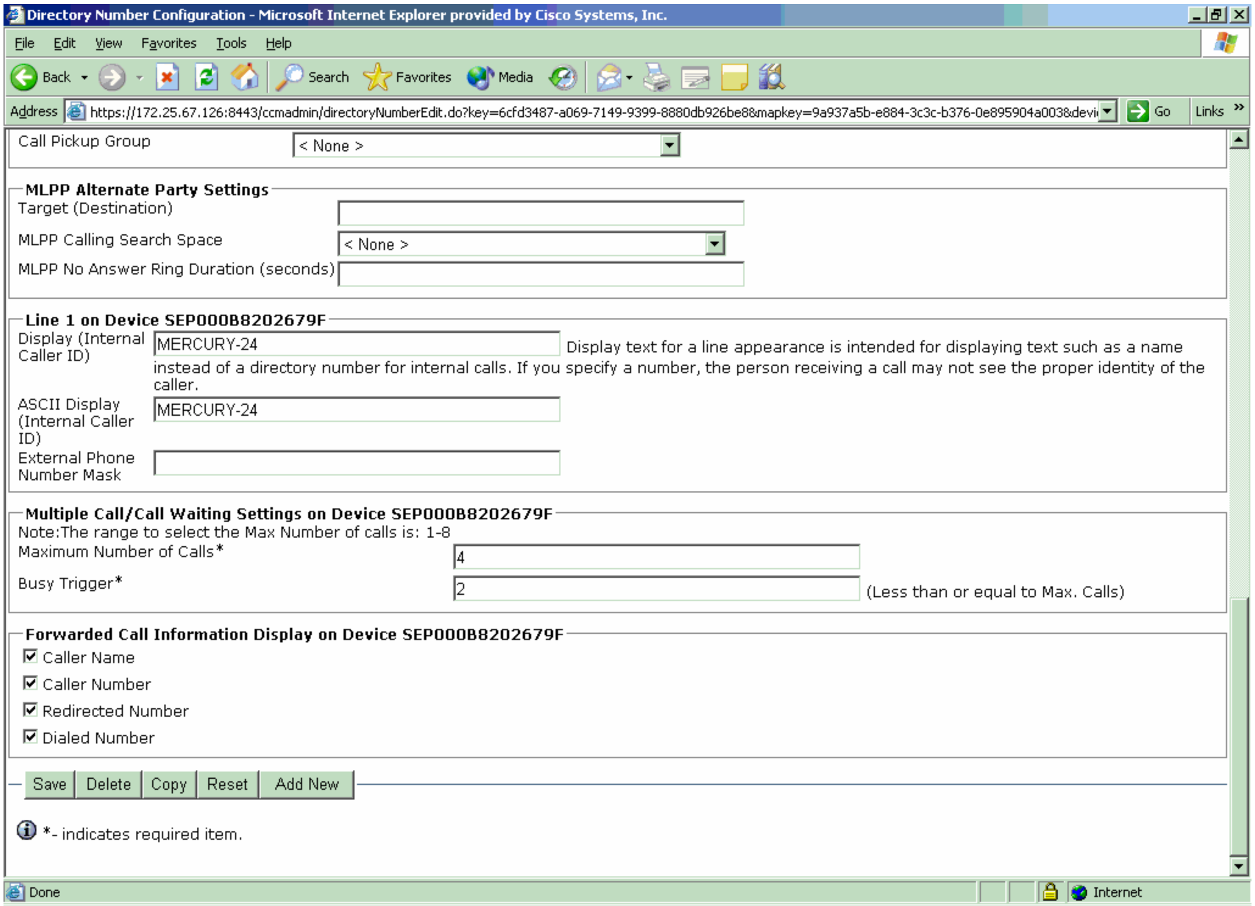
12.2.1 Grandstream Budgetone 100



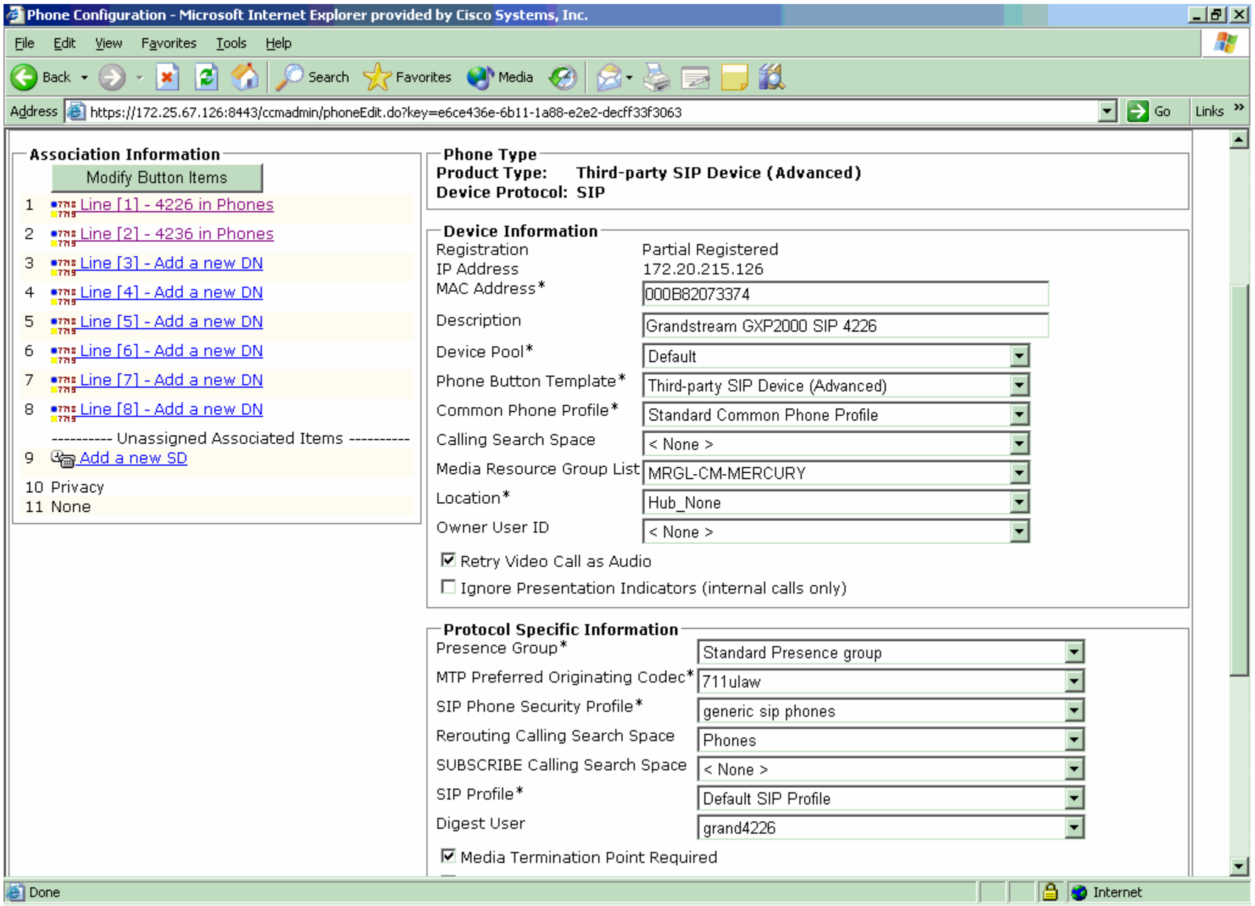

Phone Configuration - Microsoft Internet Explorer provided by Cisco Systems, Inc.
File Edit View Favorites Tools Help
Back Search Favorites Media
Address https://172.25.67.126:8443/ccmadmin/phoneEdit.do?key=5ff88d08-cc9d-9340-4e5a-13777256d8cf
Go Links
Media Resource Group List MRGL-CM-MERCURY
Location* Hub_None
Owner User ID < None >
Ignore Presentation Indicators (internal calls only)
Protocol Specific Information
Presence Group* Standard Presence group
MTP Preferred Originating Codec* 711ulaw
SIP Phone Security Profile* generic sip phones
Rerouting Calling Search Space Phones
SUBSCRIBE Calling Search Space < None >
SIP Profile* Default SIP Profile
Digest User grand4224
Media Termination Point Required
Unattended Port
MLPP Information
MLPP Domain < None >
Secure Shell Information
Secure Shell User
Secure Shell Password
Save Delete Copy Reset Add New
*- indicates required item.
- Device reset is not required for changes to Packet Capture Mode and Packet Capture Duration.
Done
Internet

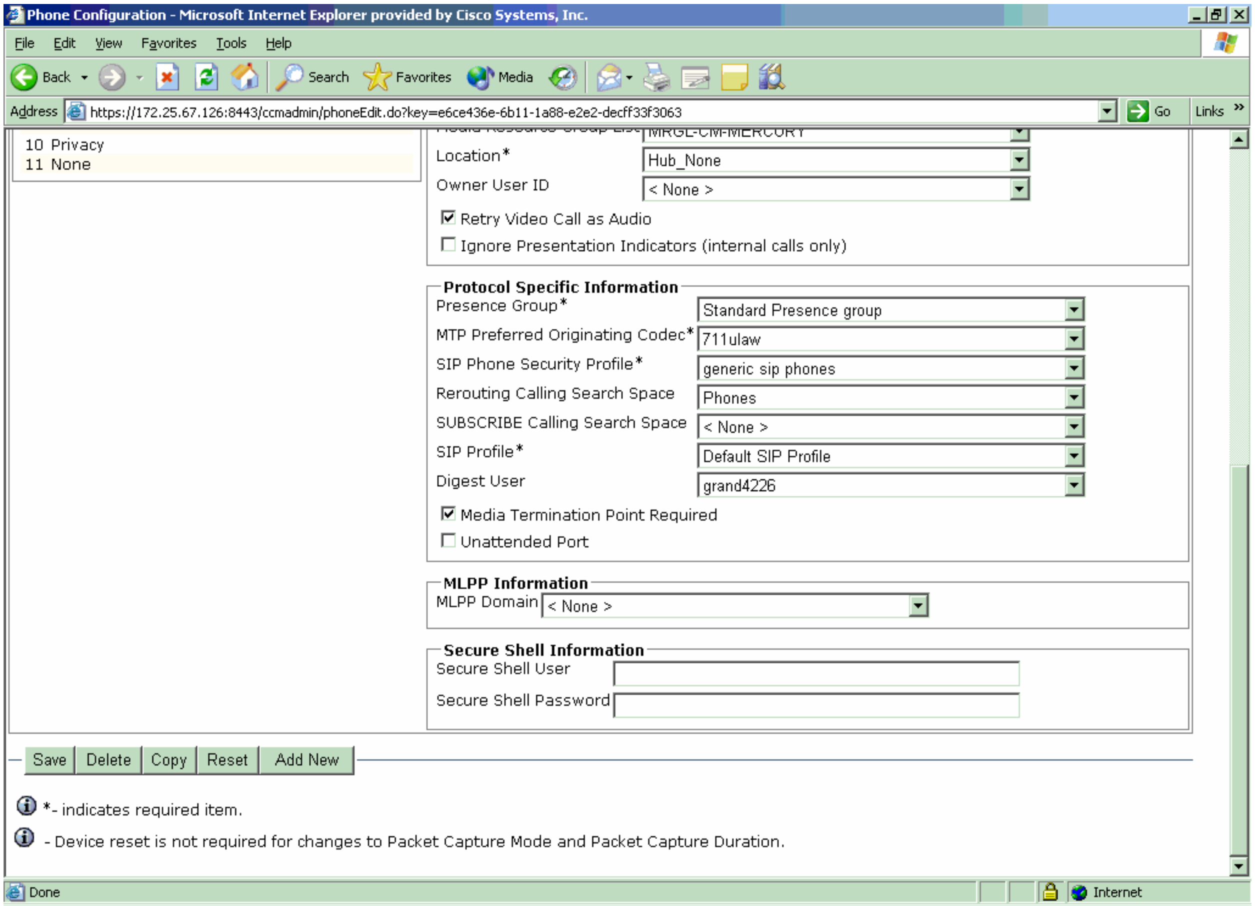

Directory Number Configuration - Microsoft Internet Explorer provided by Cisco Systems, Inc.
File Edit View Favorites Tools Help
Back Search Favorites Media
Address https://172.25.67.126:8443/ccmadmin/directoryNumberEdit.do?key=6cfd3487-a069-7149-9399-8880db926be8&mapkey=9a937a5b-e884-3c3c-b376-0e895904a003&devi
Go Links
Directory Number Information
Directory Number*
4224
Route Partition
Phones
Description
Alerting Name
MERCURY-24(A)
ASCII Alerting Name
MERCURY-24(A)
Associated Devices
SEP000B8202679F
Edit Device
Edit Line Appearance
Dissociate Devices
Directory Number Settings
Voice Mail Profile
< None >
(Choose <None> to use system default)
Calling Search Space
Phones
Presence Group*
Standard Presence group
AAR Group
< None >
User Hold Audio Source
1-SampleAudioSource
Network Hold Audio Source
< None >
Call Forward and Call Pickup Settings
Voice Mail
Destination
Calling Search Space
Forward All
or
Phones
Secondary Calling Search Space for Forward All
< None >
Find
Done
Internet

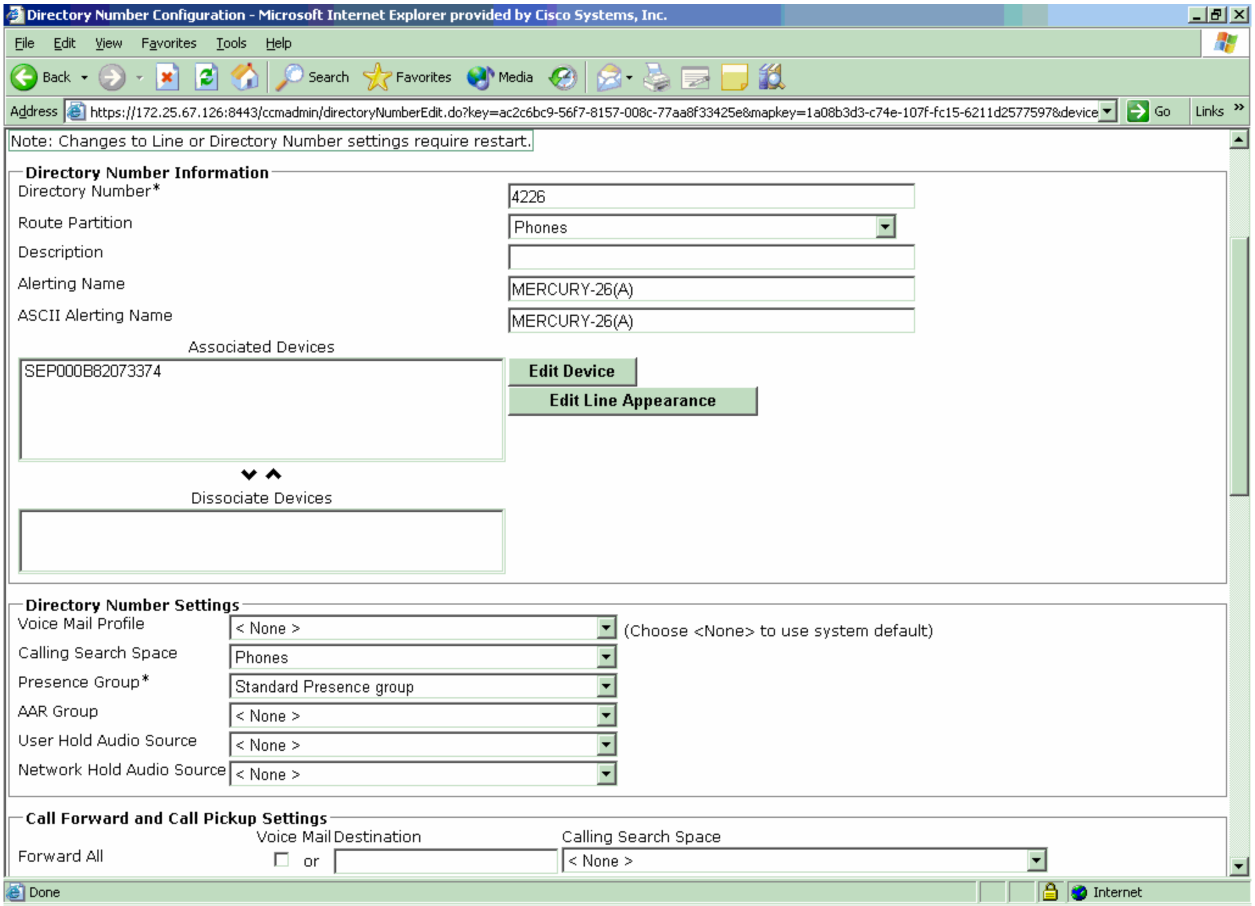

Directory Number Configuration - Microsoft Internet Explorer provided by Cisco Systems, Inc.
File Edit View Favorites Tools Help
Back Search Favorites Media
Address https://172.25.67.126:8443/ccmadmin/directoryNumberEdit.do?key=6cfd3487-a069-7149-9399-8880db926be8&mapkey=9a937a5b-e884-3c3c-b376-0e895904a003&devi
Go Links
Forward Busy Internal or Phones
Forward Busy External or Phones
Forward No Answer Internal or Phones
Forward No Answer External or Phones
Forward No Coverage Internal or < None >
Forward No Coverage External or < None >
Forward on CTI Failure or < None >
No Answer Ring Duration (seconds)
Call Pickup Group < None >
MLPP Alternate Party Settings
Target (Destination)
MLPP Calling Search Space < None >
MLPP No Answer Ring Duration (seconds)
Line 1 on Device SEP000B8202679F
Display (Internal Caller ID) MERCURY-24 Display text for a line appearance is intended for displaying text such as a name instead of a directory number for internal calls. If you specify a number, the person receiving a call may not see the proper identity of the caller.
ASCII Display (Internal Caller ID) MERCURY-24
External Phone Number Mask
Multiple Call/Call Waiting Settings on Device SEP000B8202679F
Note:The range to select the Max Number of calls is: 1-8
Maximum Number of Calls* 4
Busy Trigger* 2 (Less than or equal to Max. Calls)
Done
Internet

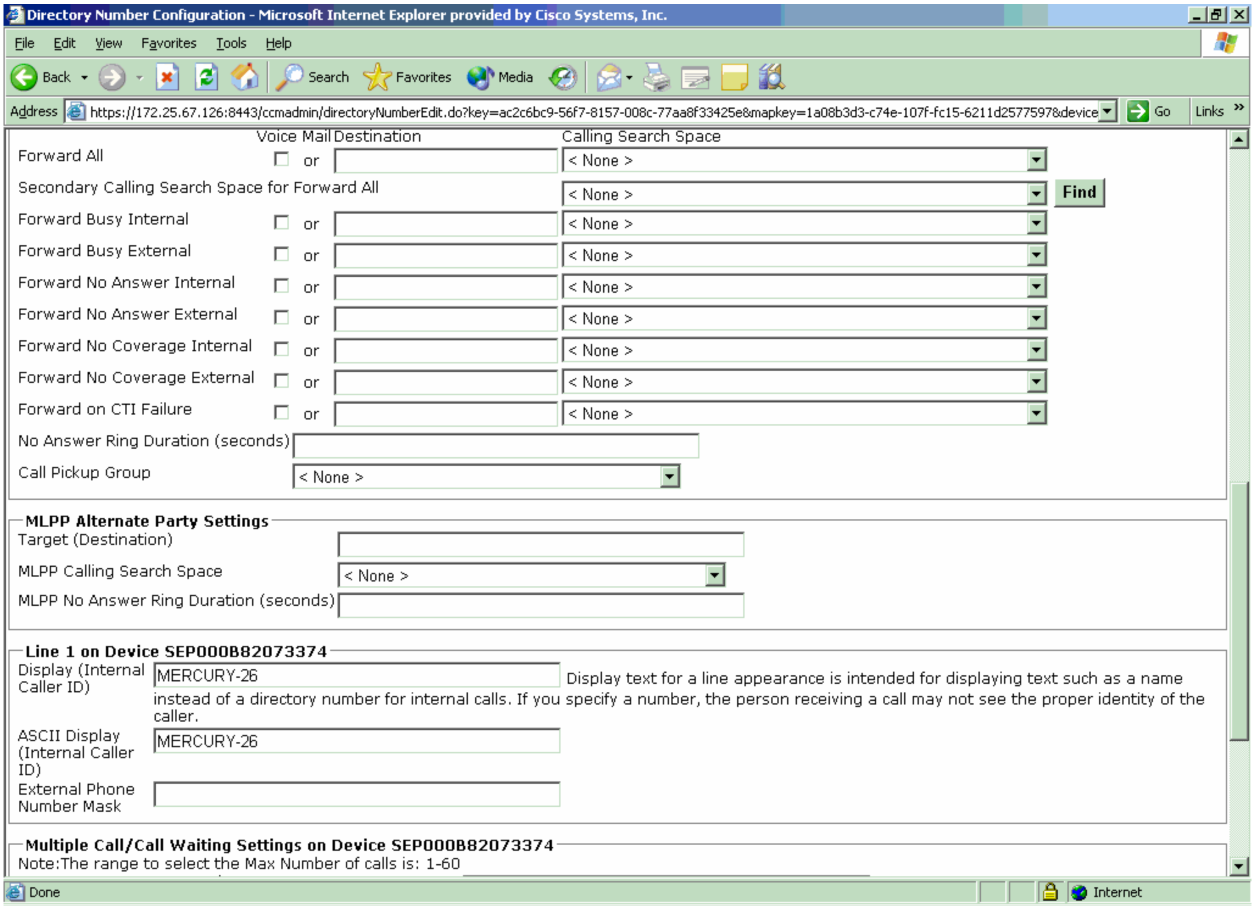

Directory Number Configuration - Microsoft Internet Explorer provided by Cisco Systems, Inc.
File Edit View Favorites Tools Help
Back Search Favorites Media
Address https://172.25.67.126:8443/ccmadmin/directoryNumberEdit.do?key=6cfd3487-a069-7149-9399-8880db926be8&mapkey=9a937a5b-e884-3c3c-b376-0e895904a003&devi
Go Links
Call Pickup Group
< None >
MLPP Alternate Party Settings
Target (Destination)
MLPP Calling Search Space
< None >
MLPP No Answer Ring Duration (seconds)
Line 1 on Device SEP000B8202679F
Display (Internal Caller ID)
MERCURY-24
Display text for a line appearance is intended for displaying text such as a name instead of a directory number for internal calls. If you specify a number, the person receiving a call may not see the proper identity of the caller.
ASCII Display (Internal Caller ID)
MERCURY-24
External Phone Number Mask
Multiple Call/Call Waiting Settings on Device SEP000B8202679F
Note:The range to select the Max Number of calls is: 1-8
Maximum Number of Calls*
4
Busy Trigger*
2
(Less than or equal to Max. Calls)
Forwarded Call Information Display on Device SEP000B8202679F
Caller Name
Caller Number
Redirected Number
Dialed Number
Save Delete Copy Reset Add New
*- indicates required item.
Done
Internet

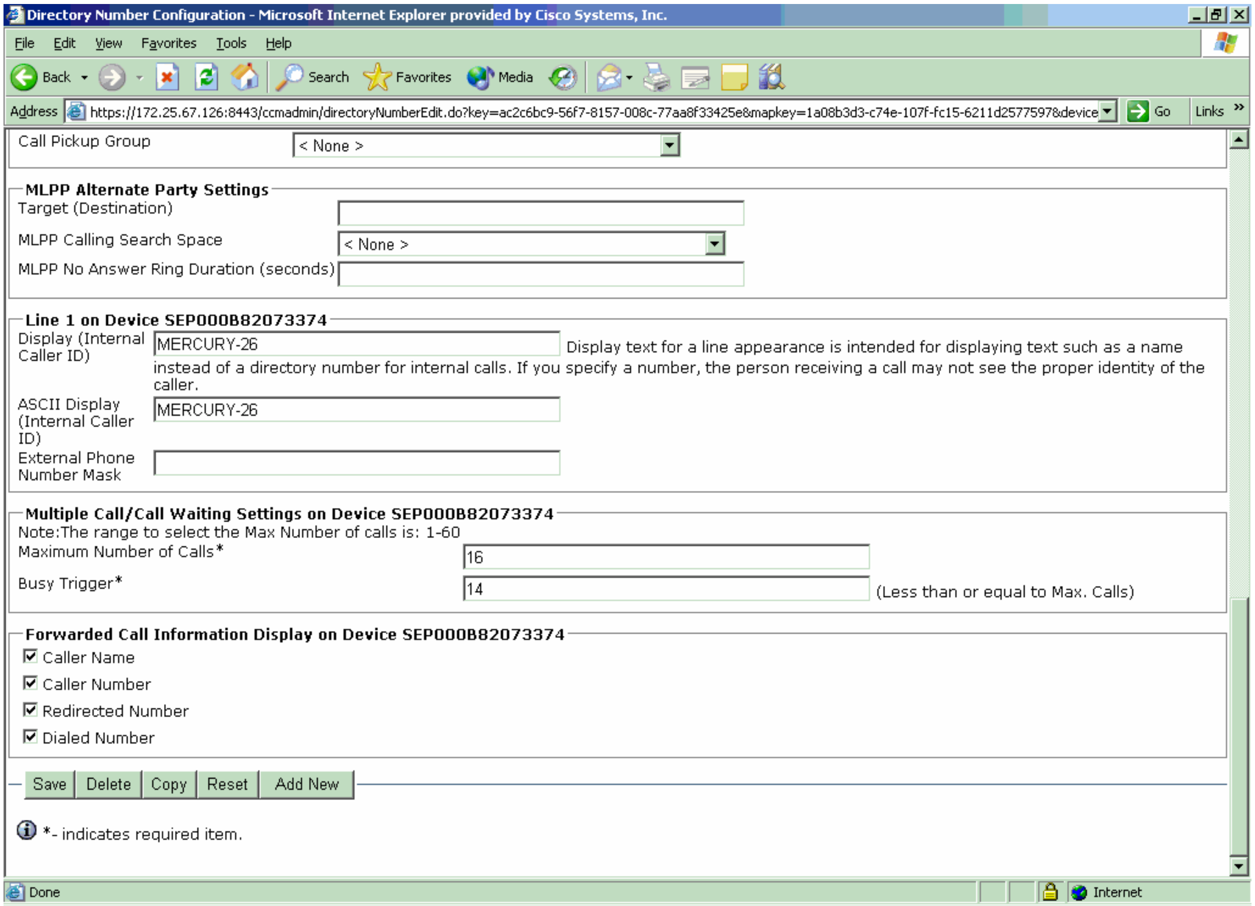
12.2.2 Grandstream GXP-2000



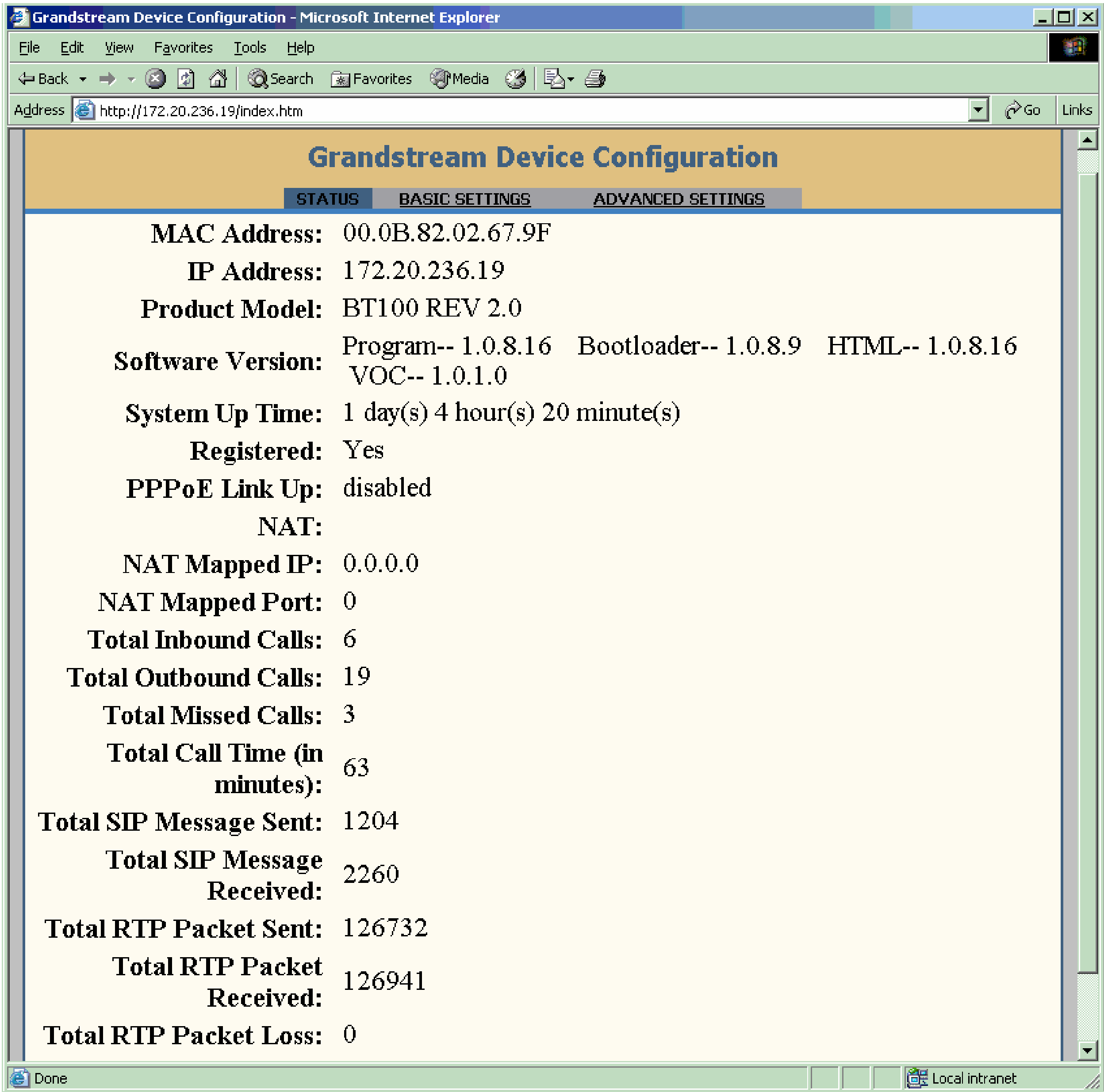

Phone Configuration - Microsoft Internet Explorer provided by Cisco Systems, Inc.
File Edit View Favorites Tools Help
Back Search Favorites Media
Address https://172.25.67.126:8443/ccmadmin/phoneEdit.do?key=e6ce436e-6b11-1a88-e2e2-decff33f3063
Go Links
10 Privacy
11 None
Location* Hub_None
Owner User ID < None >
Retry Video Call as Audio
Ignore Presentation Indicators (internal calls only)
Protocol Specific Information
Presence Group* Standard Presence group
MTP Preferred Originating Codec* 711ulaw
SIP Phone Security Profile* generic sip phones
Rerouting Calling Search Space Phones
SUBSCRIBE Calling Search Space < None >
SIP Profile* Default SIP Profile
Digest User grand4226
Media Termination Point Required
Unattended Port
MLPP Information
MLPP Domain < None >
Secure Shell Information
Secure Shell User
Secure Shell Password
Save Delete Copy Reset Add New
*- indicates required item.
- Device reset is not required for changes to Packet Capture Mode and Packet Capture Duration.
Done
Internet

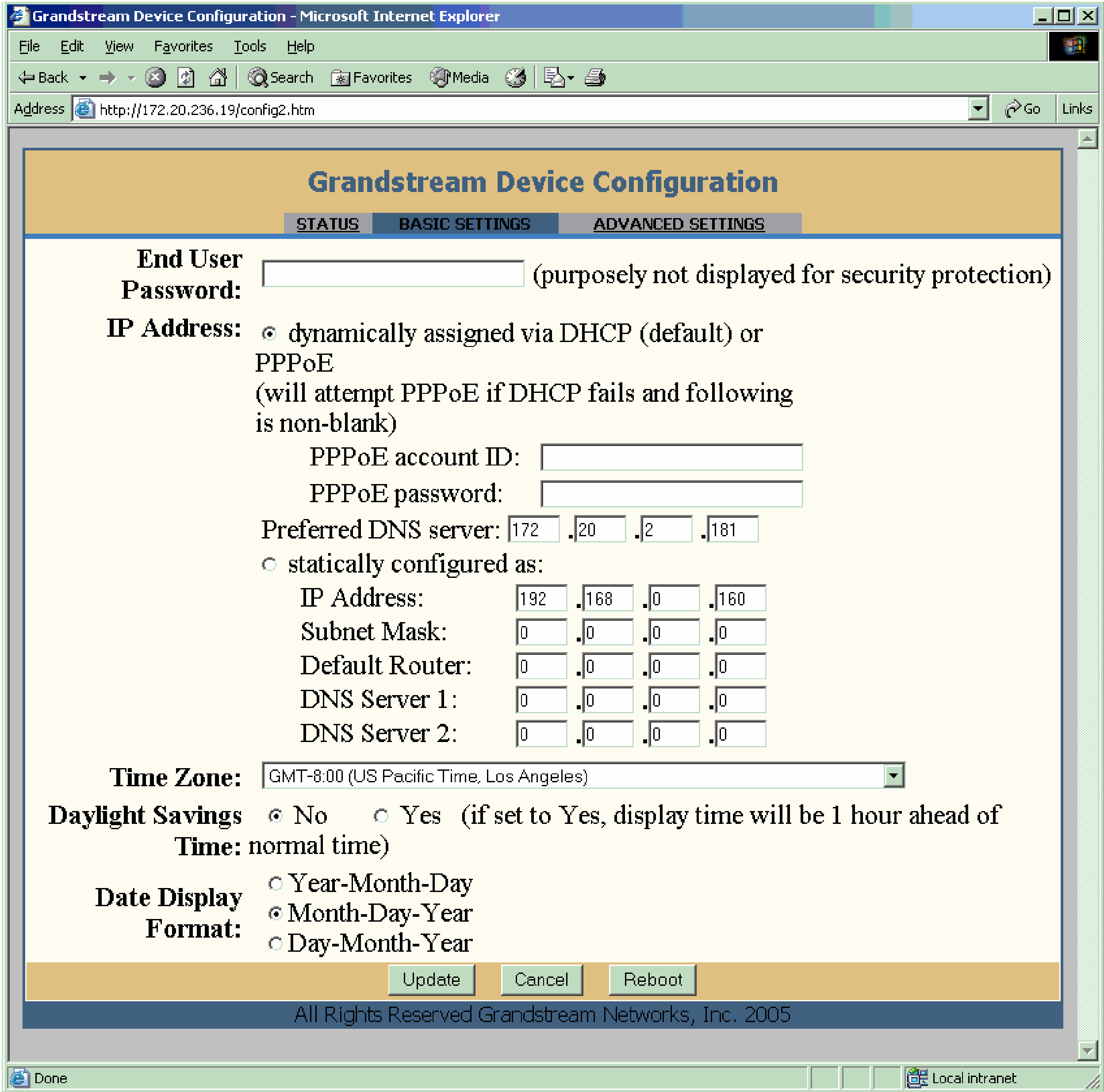

Directory Number Configuration - Microsoft Internet Explorer provided by Cisco Systems, Inc.
File Edit View Favorites Tools Help
Back Search Favorites Media
Address https://172.25.67.126:8443/ccmadmin/directoryNumberEdit.do?key=ac2c6bc9-56f7-8157-008c-77aa8f33425e&mapkey=1a08b3d3-c74e-107f-fc15-6211d2577597&device
Go Links
Note: Changes to Line or Directory Number settings require restart.
Directory Number Information
Directory Number* 4226
Route Partition Phones
Description
Alerting Name MERCURY-26(A)
ASCII Alerting Name MERCURY-26(A)
Associated Devices
SEP000B82073374
Edit Device
Edit Line Appearance
Dissociate Devices
Directory Number Settings
Voice Mail Profile < None > (Choose <None> to use system default)
Calling Search Space Phones
Presence Group* Standard Presence group
AAR Group < None >
User Hold Audio Source < None >
Network Hold Audio Source < None >
Call Forward and Call Pickup Settings
Voice Mail Destination
Calling Search Space
Forward All or < None >
Done
Internet

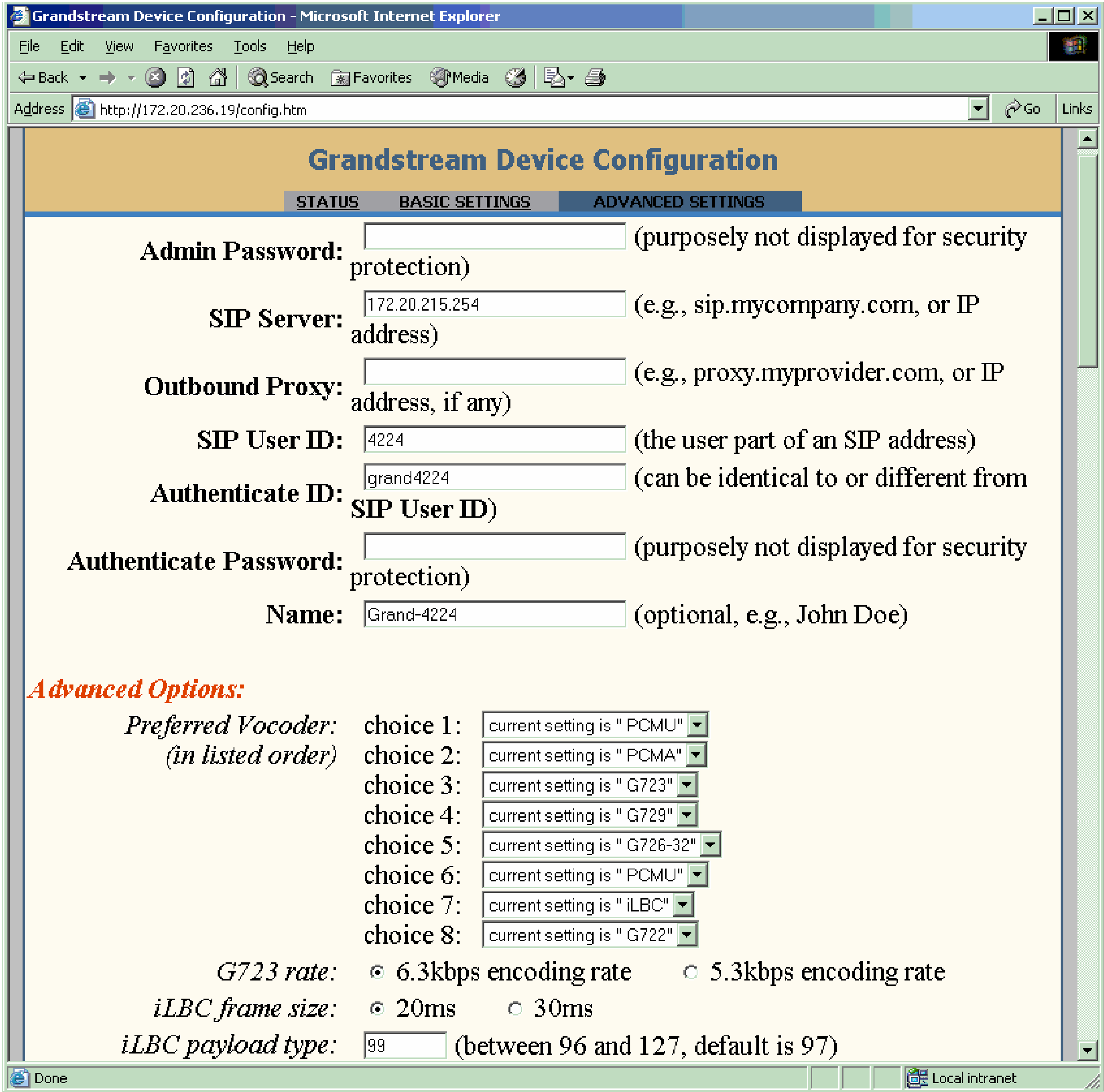

Directory Number Configuration - Microsoft Internet Explorer provided by Cisco Systems, Inc.
File Edit View Favorites Tools Help
Back Search Favorites Media
Address https://172.25.67.126:8443/ccmadmin/directoryNumberEdit.do?key=ac2c6bc9-56f7-8157-008c-77aa8f33425e&mapkey=1a08b3d3-c74e-107f-fc15-6211d2577597&device
Go Links
Voice Mail Destination
Calling Search Space
Forward All
or
< None >
Secondary Calling Search Space for Forward All
Find
Forward Busy Internal
Forward Busy External
Forward No Answer Internal
Forward No Answer External
Forward No Coverage Internal
Forward No Coverage External
Forward on CTI Failure
No Answer Ring Duration (seconds)
Call Pickup Group
MLPP Alternate Party Settings
Target (Destination)
MLPP Calling Search Space
MLPP No Answer Ring Duration (seconds)
Line 1 on Device SEP000B82073374
Display (Internal Caller ID)
MERCURY-26
Display text for a line appearance is intended for displaying text such as a name instead of a directory number for internal calls. If you specify a number, the person receiving a call may not see the proper identity of the caller.
ASCII Display (Internal Caller ID)
MERCURY-26
External Phone Number Mask
Multiple Call/Call Waiting Settings on Device SEP000B82073374
Note:The range to select the Max Number of calls is: 1-60
Done
Internet

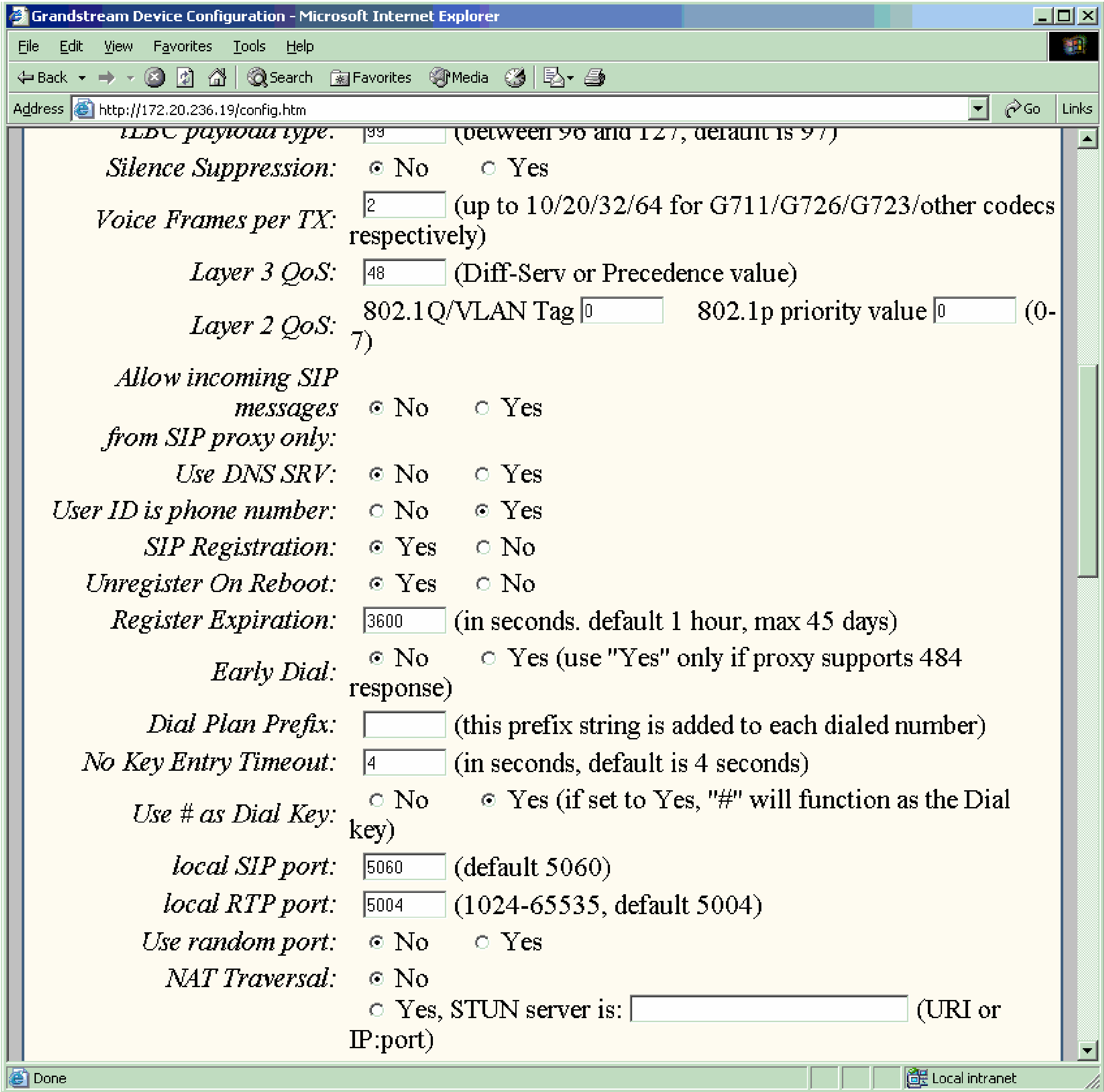

Directory Number Configuration - Microsoft Internet Explorer provided by Cisco Systems, Inc.
File Edit View Favorites Tools Help
Back Search Favorites Media
Address https://172.25.67.126:8443/ccmadmin/directoryNumberEdit.do?key=ac2c6bc9-56f7-8157-008c-77aa8f33425e&mapkey=1a08b3d3-c74e-107f-fc15-6211d25775978device
Go Links
Call Pickup Group
< None >
MLPP Alternate Party Settings
Target (Destination)
MLPP Calling Search Space
< None >
MLPP No Answer Ring Duration (seconds)
Line 1 on Device SEP000B82073374
Display (Internal Caller ID)
MERCURY-26
Display text for a line appearance is intended for displaying text such as a name instead of a directory number for internal calls. If you specify a number, the person receiving a call may not see the proper identity of the caller.
ASCII Display (Internal Caller ID)
MERCURY-26
External Phone Number Mask
Multiple Call/Call Waiting Settings on Device SEP000B82073374
Note:The range to select the Max Number of calls is: 1-60
Maximum Number of Calls*
16
Busy Trigger*
14
(Less than or equal to Max. Calls)
Forwarded Call Information Display on Device SEP000B82073374
Caller Name
Caller Number
Redirected Number
Dialed Number
Save
Delete
Copy
Reset
Add New
*- indicates required item.
Done
Internet

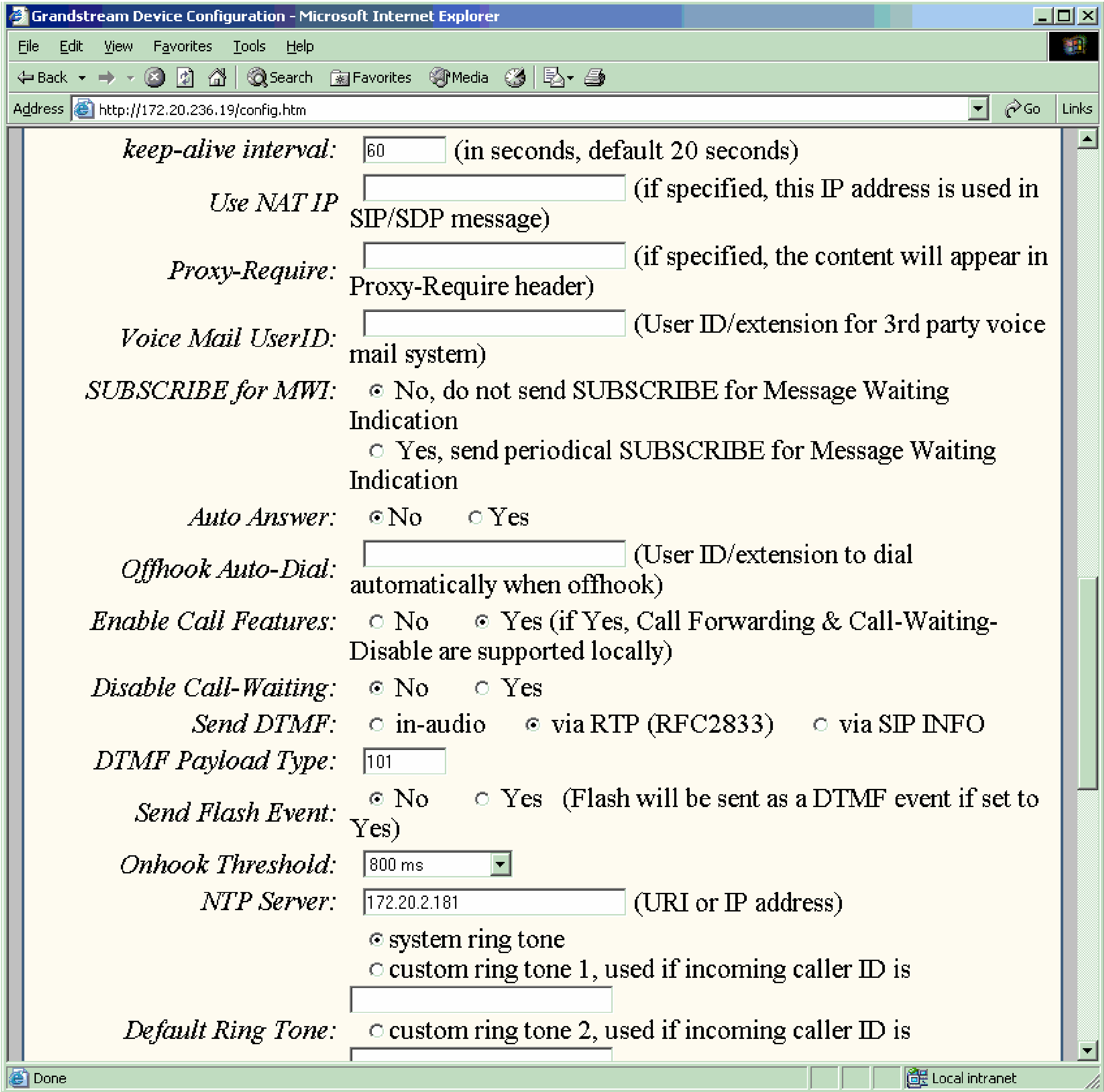
12.3 GrandStream Budgetone 100 SIP phone webpage Configuration



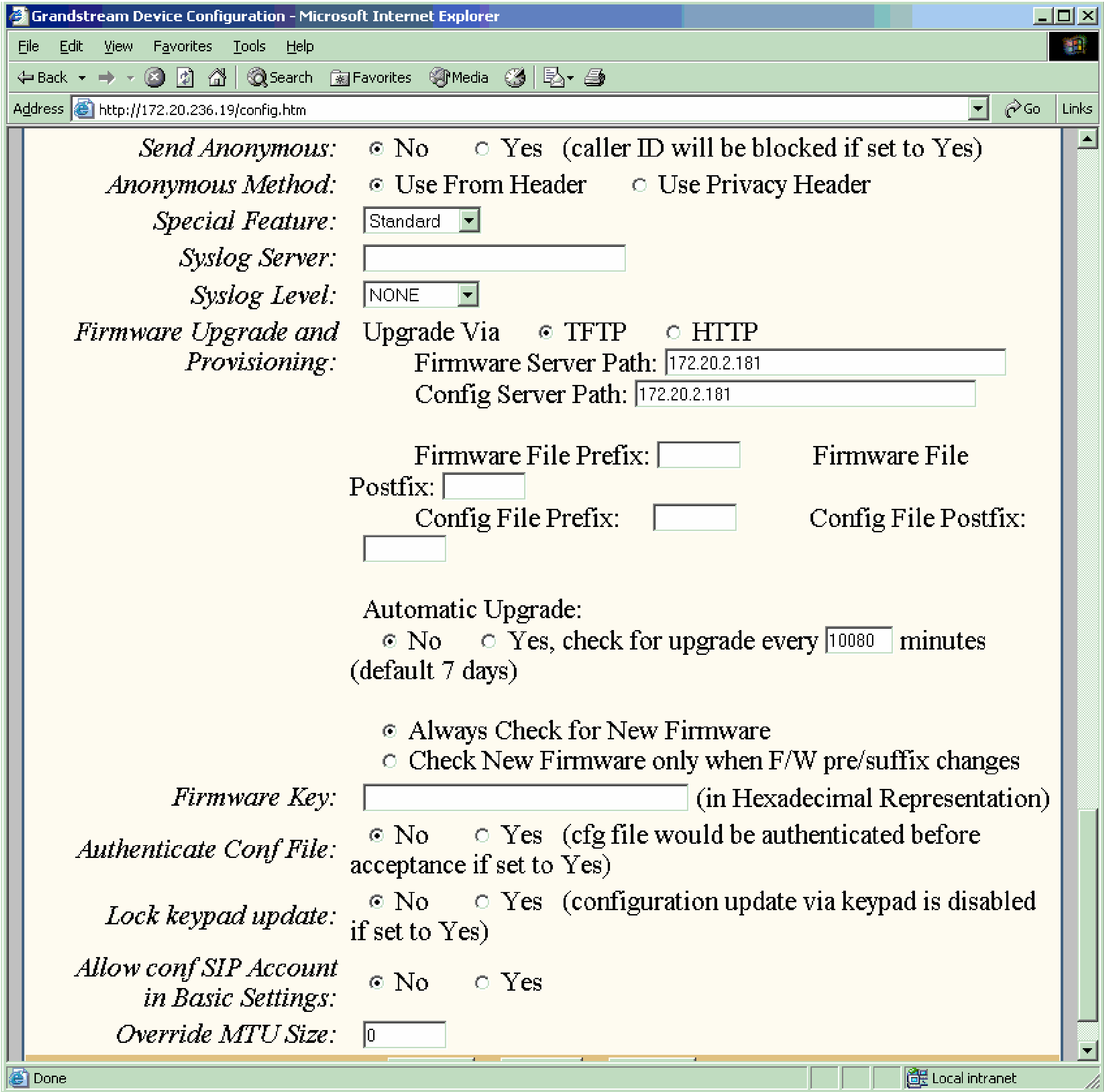

Grandstream Device Configuration - Microsoft Internet Explorer
File Edit View Favorites Tools Help
Back Search Favorites Media
Address http://172.20.236.19/config2.htm
Go Links
Grandstream Device Configuration
STATUS BASIC SETTINGS ADVANCED SETTINGS
End User Password: (purposely not displayed for security protection)
IP Address: dynamically assigned via DHCP (default) or PPPoE
(will attempt PPPoE if DHCP fails and following is non-blank)
PPPoE account ID:
PPPoE password:
Preferred DNS server: 172 .20 .2 .181
statically configured as:
IP Address: 192 .168 .0 .160
Subnet Mask: 0 .0 .0 .0
Default Router: 0 .0 .0 .0
DNS Server 1: 0 .0 .0 .0
DNS Server 2: 0 .0 .0 .0
Time Zone: GMT-8:00 (US Pacific Time, Los Angeles)
Daylight Savings Time: No Yes (if set to Yes, display time will be 1 hour ahead of normal time)
Date Display Format: Year-Month-Day
Month-Day-Year
Day-Month-Year
Update Cancel Reboot
All Rights Reserved Grandstream Networks, Inc. 2005
Done
Local intranet


Grandstream Device Configuration - Microsoft Internet Explorer
File Edit View Favorites Tools Help
Back Search Favorites Media
Address http://172.20.236.19/config.htm
Go Links
Grandstream Device Configuration
STATUS BASIC SETTINGS ADVANCED SETTINGS
Admin Password: (purposely not displayed for security protection)
SIP Server: 172.20.215.254 (e.g., sip.mycompany.com, or IP address)
Outbound Proxy: (e.g., proxy.myprovider.com, or IP address, if any)
SIP User ID: 4224 (the user part of an SIP address)
Authenticate ID: grand4224 (can be identical to or different from SIP User ID)
Authenticate Password: (purposely not displayed for security protection)
Name: Grand-4224 (optional, e.g., John Doe)
Advanced Options:
Preferred Vocoder: (in listed order)
choice 1: current setting is " PCMU"
choice 2: current setting is " PCMA"
choice 3: current setting is " G723"
choice 4: current setting is " G729"
choice 5: current setting is " G726-32"
choice 6: current setting is " PCMU"
choice 7: current setting is " iLBC"
choice 8: current setting is " G722"
G723 rate: 6.3kbps encoding rate 5.3kbps encoding rate
iLBC frame size: 20ms 30ms
iLBC payload type: 99 (between 96 and 127, default is 97)
Done
Local intranet

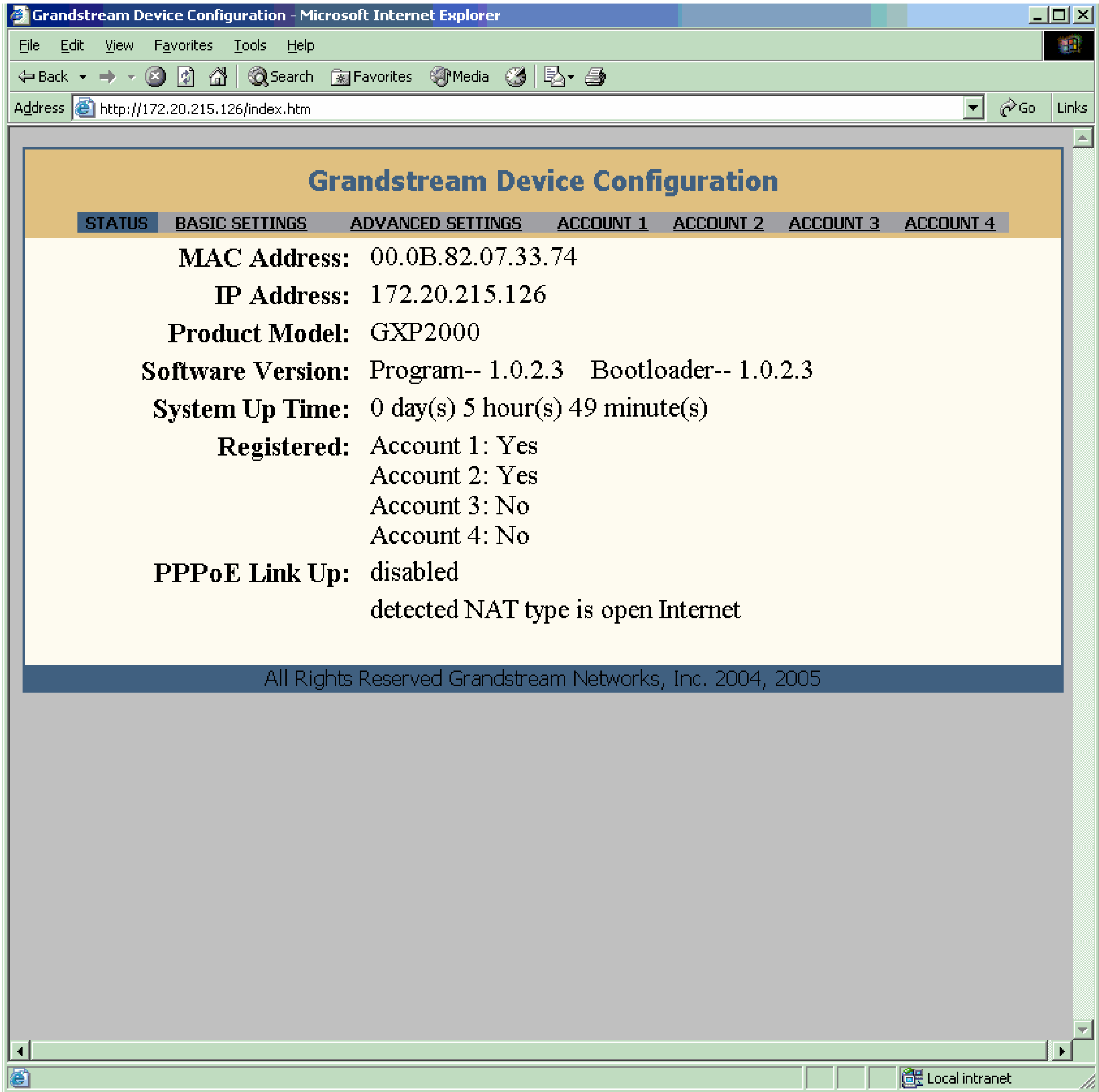

Grandstream Device Configuration - Microsoft Internet Explorer
File Edit View Favorites Tools Help
Back Search Favorites Media
Address http://172.20.236.19/config.htm
Go Links
Silence Suppression: No Yes
Voice Frames per TX: 2 (up to 10/20/32/64 for G711/G726/G723/other codecs respectively)
Layer 3 QoS: 48 (Diff-Serv or Precedence value)
Layer 2 QoS: 802.1Q/VLAN Tag 0 802.1p priority value 0 (0-7)
Allow incoming SIP messages from SIP proxy only: No Yes
Use DNS SRV: No Yes
User ID is phone number: No Yes
SIP Registration: Yes No
Unregister On Reboot: Yes No
Register Expiration: 3600 (in seconds. default 1 hour, max 45 days)
Early Dial: No Yes (use "Yes" only if proxy supports 484 response)
Dial Plan Prefix: (this prefix string is added to each dialed number)
No Key Entry Timeout: 4 (in seconds, default is 4 seconds)
Use # as Dial Key: No Yes (if set to Yes, "#" will function as the Dial key)
local SIP port: 5060 (default 5060)
local RTP port: 5004 (1024-65535, default 5004)
Use random port: No Yes
NAT Traversal: No
Yes, STUN server is: (URI or IP:port)
Done
Local intranet

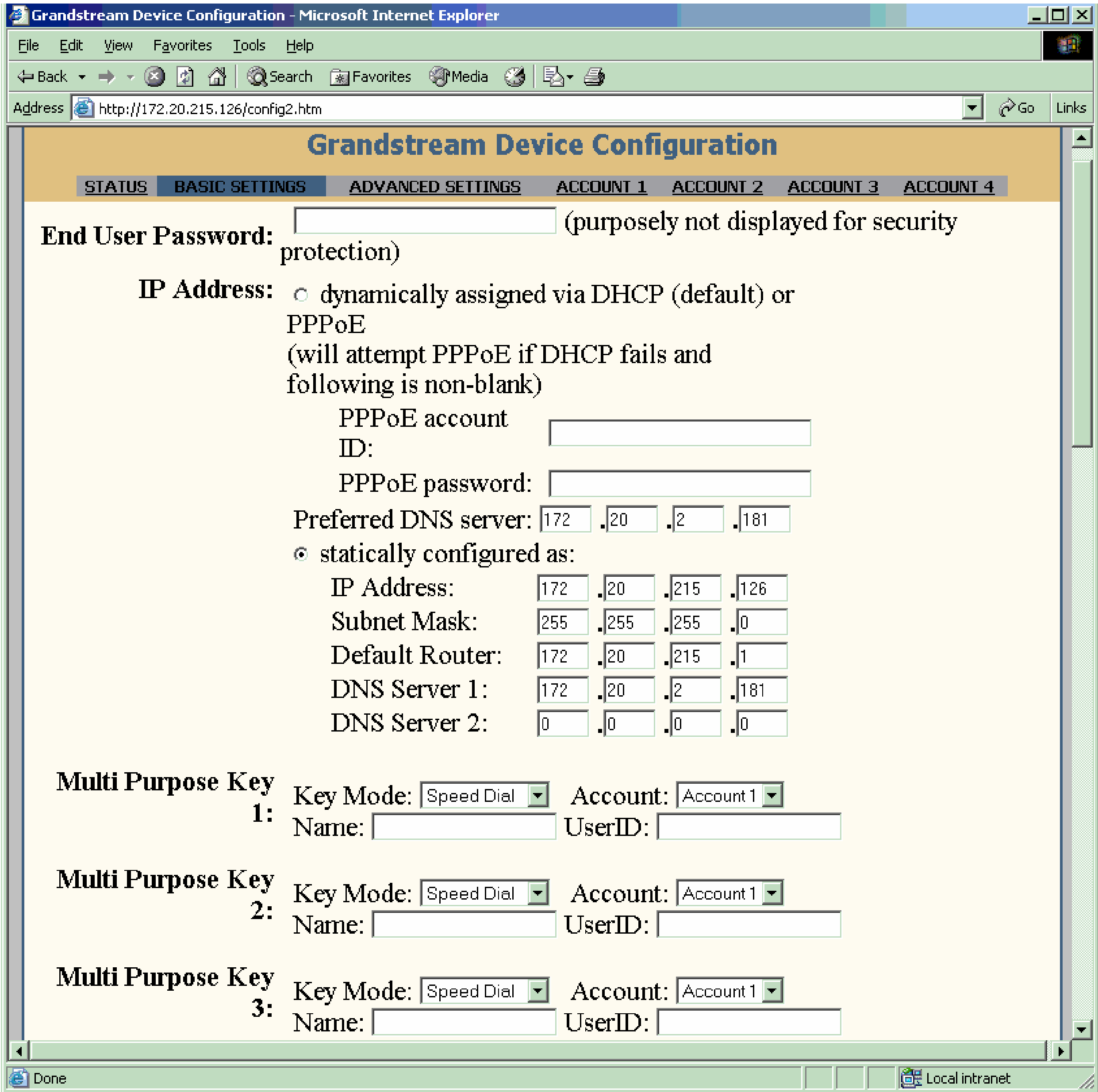


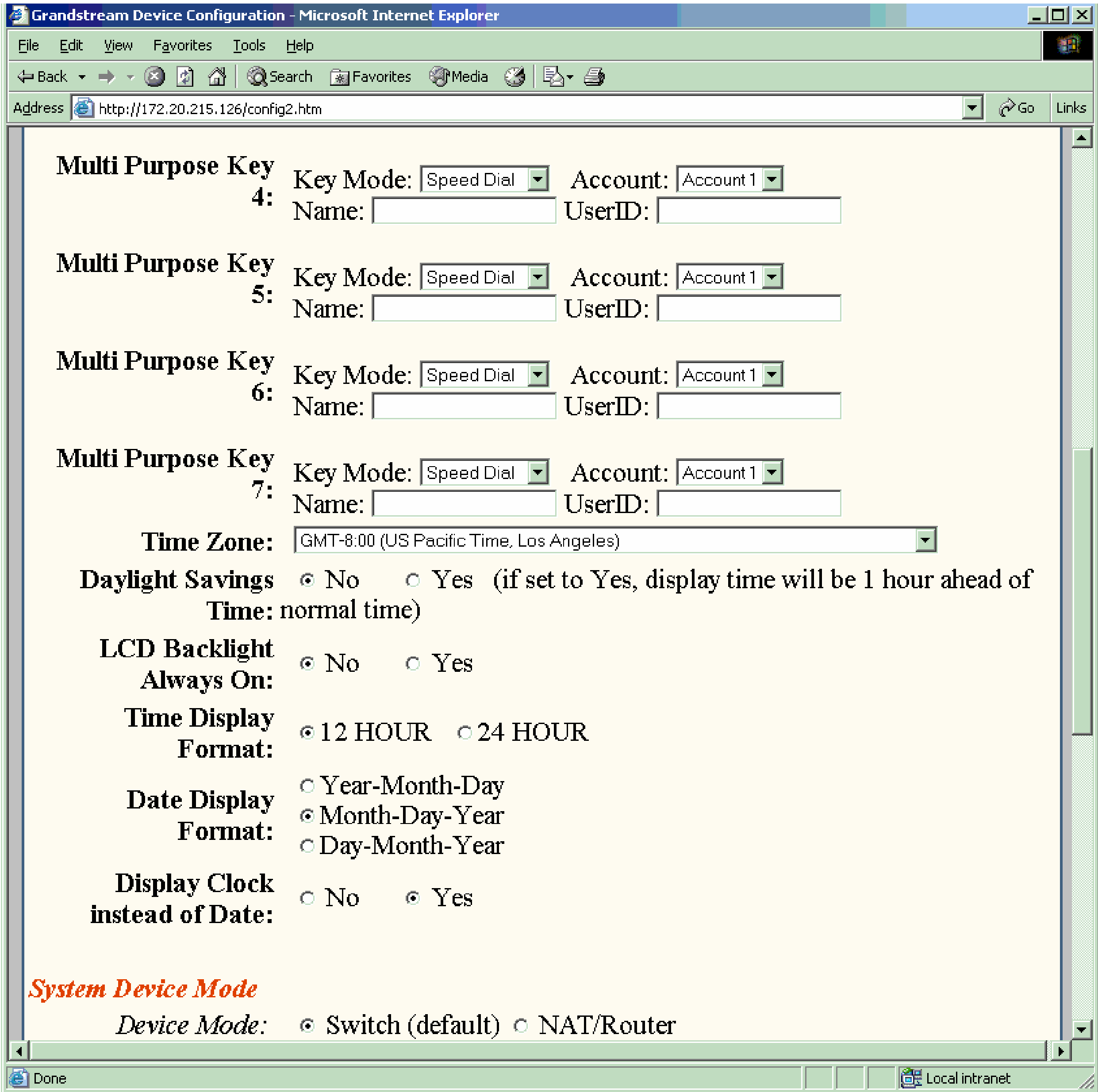
Grandstream Device Configuration - Microsoft Internet Explorer
File Edit View Favorites Tools Help
Back Search Favorites Media
Address http://172.20.236.19/config.htm
Go Links
keep-alive interval: 60 (in seconds, default 20 seconds)
Use NAT IP (if specified, this IP address is used in SIP/SDP message)
Proxy-Require: (if specified, the content will appear in Proxy-Require header)
Voice Mail UserID: (User ID/extension for 3rd party voice mail system)
SUBSCRIBE for MWI: No, do not send SUBSCRIBE for Message Waiting Indication
Yes, send periodical SUBSCRIBE for Message Waiting Indication
Auto Answer: No Yes
Offhook Auto-Dial: (User ID/extension to dial automatically when offhook)
Enable Call Features: No Yes (if Yes, Call Forwarding & Call-Waiting-Disable are supported locally)
Disable Call-Waiting: No Yes
Send DTMF: in-audio via RTP (RFC2833) via SIP INFO
DTMF Payload Type: 101
Send Flash Event: No Yes (Flash will be sent as a DTMF event if set to Yes)
Onhook Threshold: 800 ms
NTP Server: 172.20.2.181 (URI or IP address)
system ring tone
custom ring tone 1, used if incoming caller ID is
Default Ring Tone: custom ring tone 2, used if incoming caller ID is
Done
Local intranet

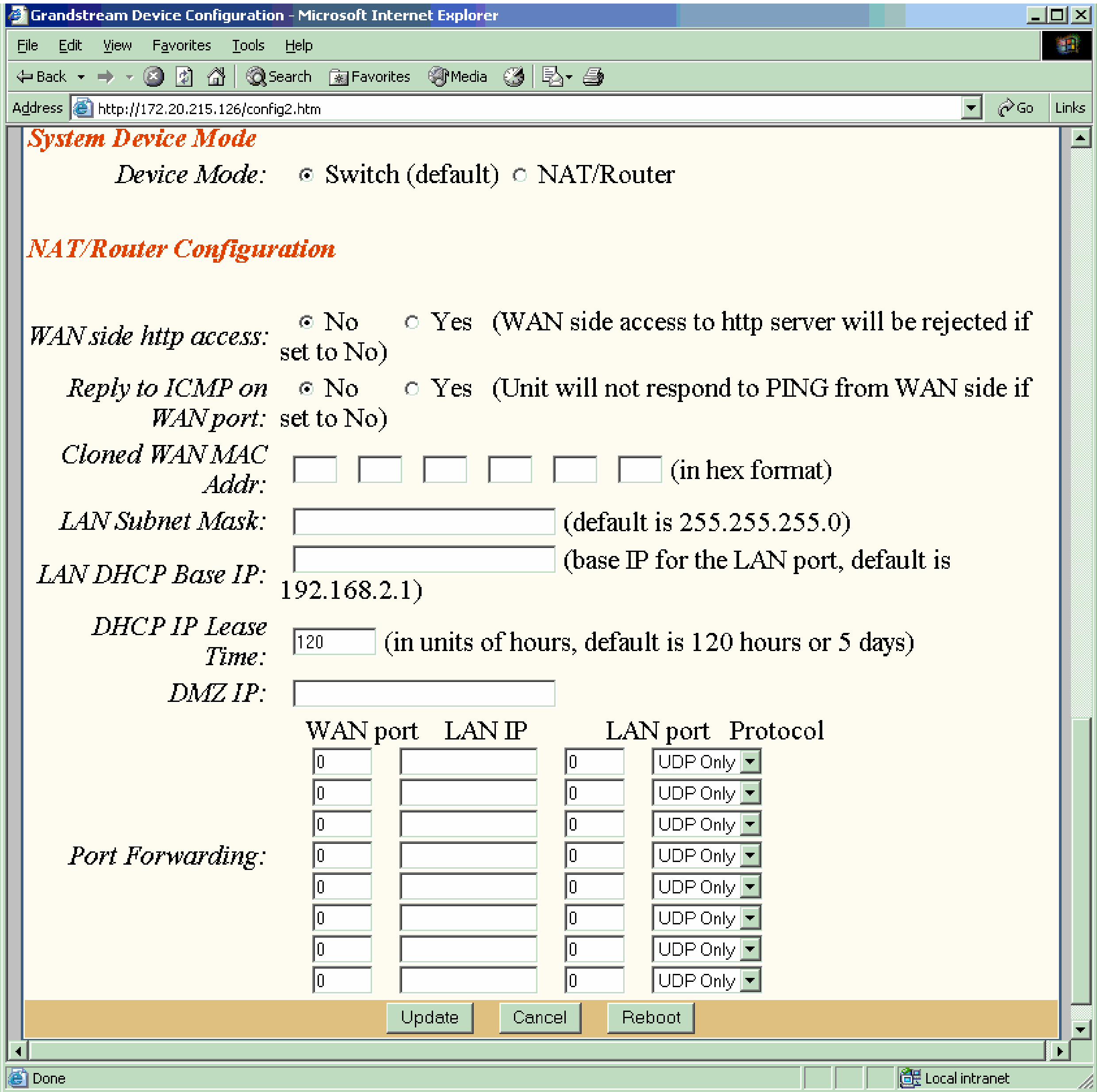

Grandstream Device Configuration - Microsoft Internet Explorer
File Edit View Favorites Tools Help
Back Search Favorites Media
Address http://172.20.236.19/config.htm
Go Links
Send Anonymous: No Yes (caller ID will be blocked if set to Yes)
Anonymous Method: Use From Header Use Privacy Header
Special Feature: Standard
Syslog Server:
Syslog Level: NONE
Firmware Upgrade and Provisioning: Upgrade Via TFTP HTTP
Firmware Server Path: 172.20.2.181
Config Server Path: 172.20.2.181
Firmware File Prefix: Firmware File Postfix:
Config File Prefix: Config File Postfix:
Automatic Upgrade:
No Yes, check for upgrade every 10080 minutes (default 7 days)
Always Check for New Firmware
Check New Firmware only when F/W pre/suffix changes
Firmware Key: (in Hexadecimal Representation)
Authenticate Conf File: No Yes (cfg file would be authenticated before acceptance if set to Yes)
Lock keypad update: No Yes (configuration update via keypad is disabled if set to Yes)
Allow conf SIP Account in Basic Settings: No Yes
Override MTU Size: 0
Done
Local intranet

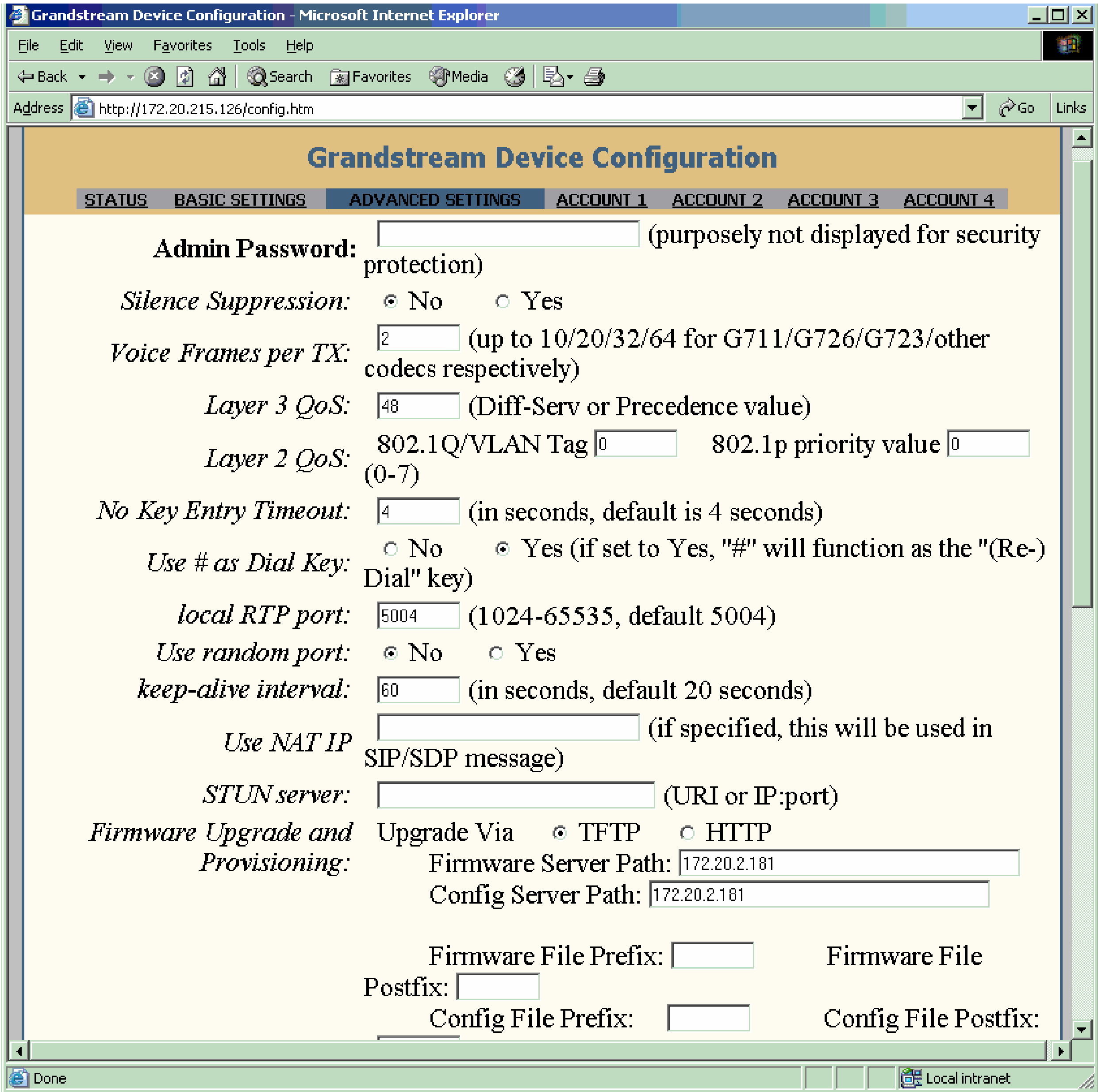
12.4 GrandStream GXP-2000 SIP phone webpage Configuration

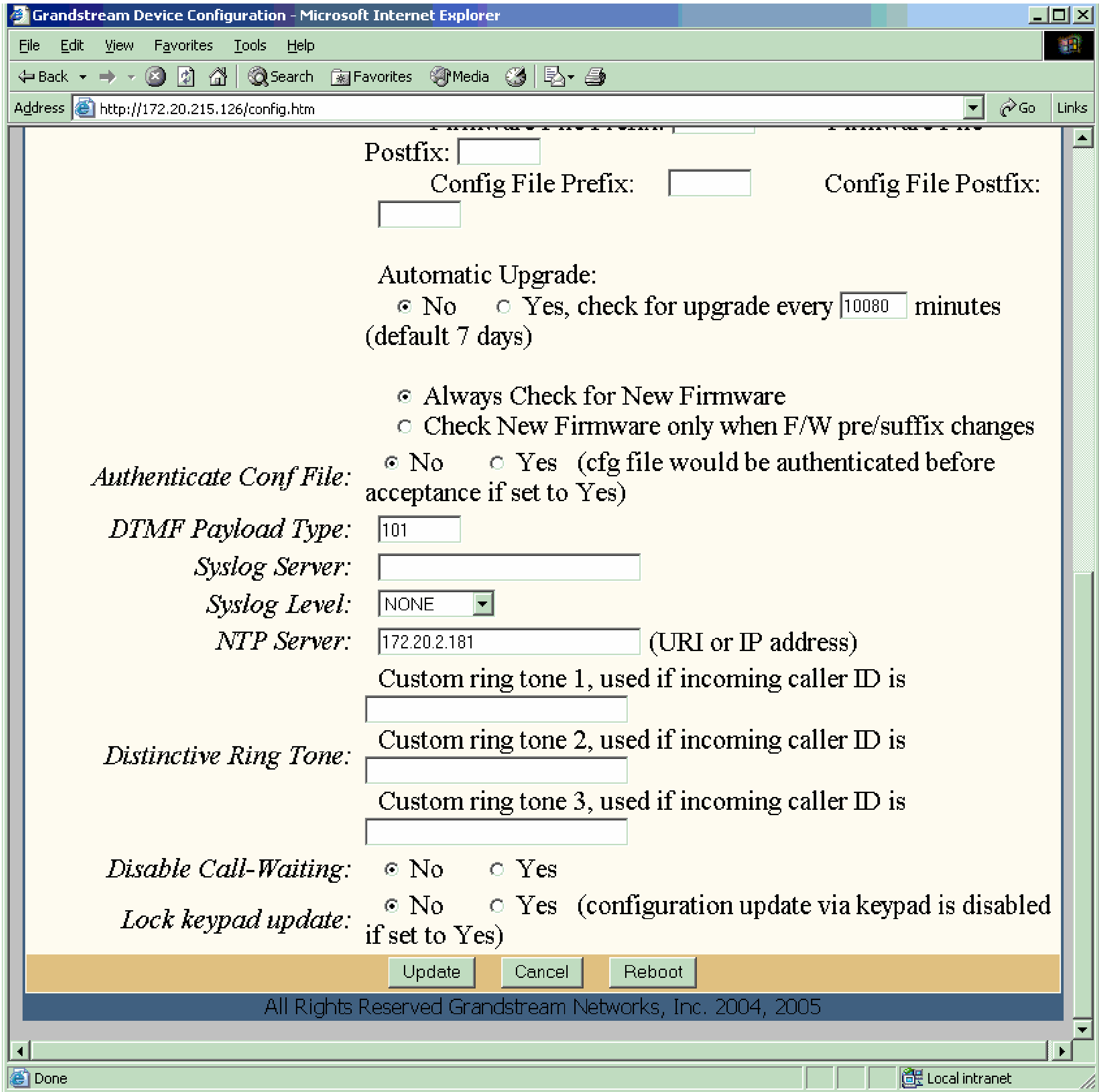


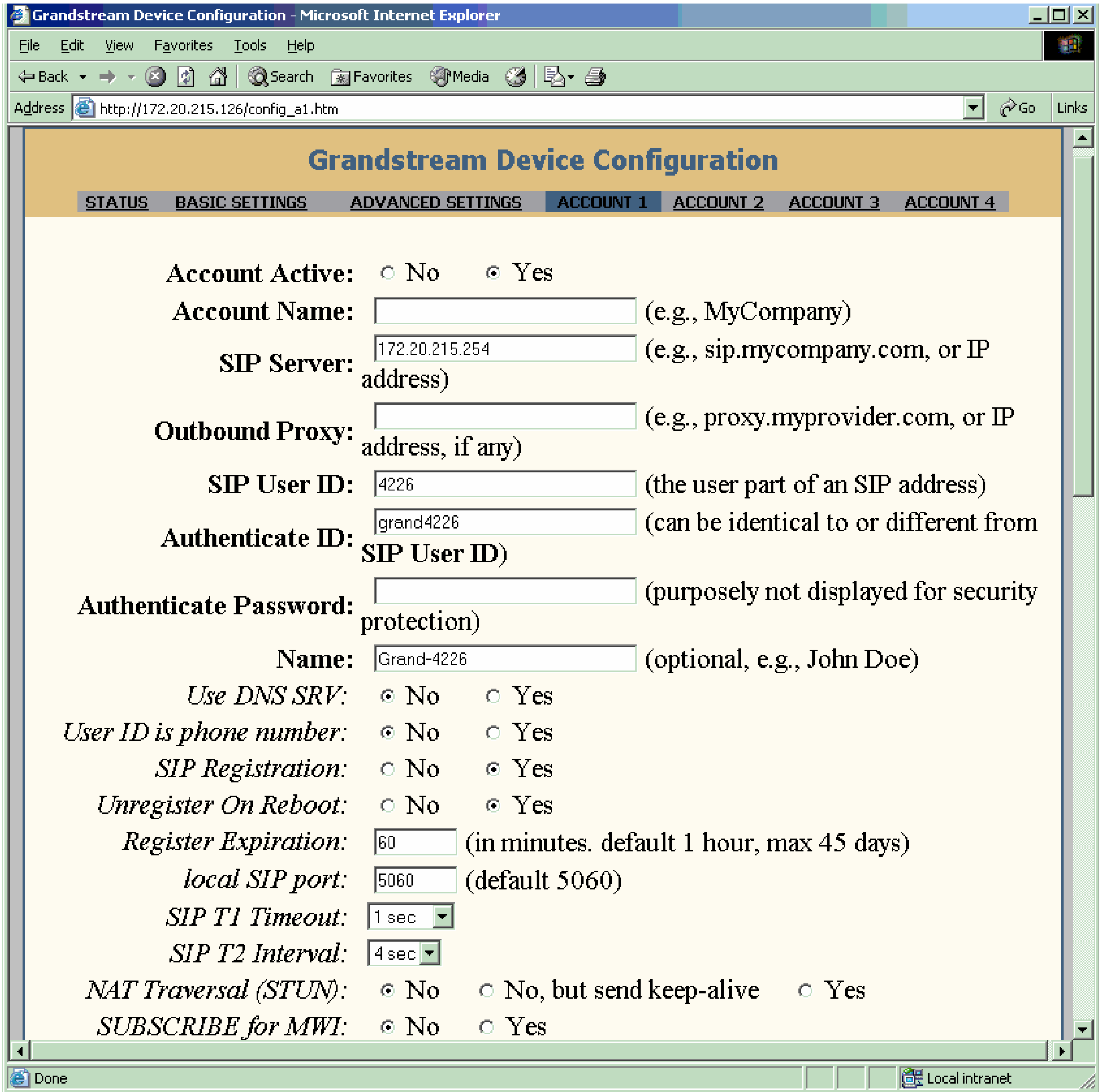

Grandstream Device Configuration - Microsoft Internet Explorer
File Edit View Favorites Tools Help
Back Search Favorites Media
Address http://172.20.215.126/config2.htm
Go Links
Grandstream Device Configuration
STATUS BASIC SETTINGS ADVANCED SETTINGS ACCOUNT 1 ACCOUNT 2 ACCOUNT 3 ACCOUNT 4
End User Password: (purposely not displayed for security protection)
IP Address: dynamically assigned via DHCP (default) or PPPoE
(will attempt PPPoE if DHCP fails and following is non-blank)
PPPoE account ID:
PPPoE password:
Preferred DNS server: 172 . 20 . 2 . 181
statically configured as:
IP Address: 172 . 20 . 215 . 126
Subnet Mask: 255 . 255 . 255 . 0
Default Router: 172 . 20 . 215 . 1
DNS Server 1: 172 . 20 . 2 . 181
DNS Server 2: 0 . 0 . 0 . 0
Multi Purpose Key 1: Key Mode: Speed Dial Account: Account 1
Name: UserID:
Multi Purpose Key 2: Key Mode: Speed Dial Account: Account 1
Name: UserID:
Multi Purpose Key 3: Key Mode: Speed Dial Account: Account 1
Name: UserID:
Done
Local intranet

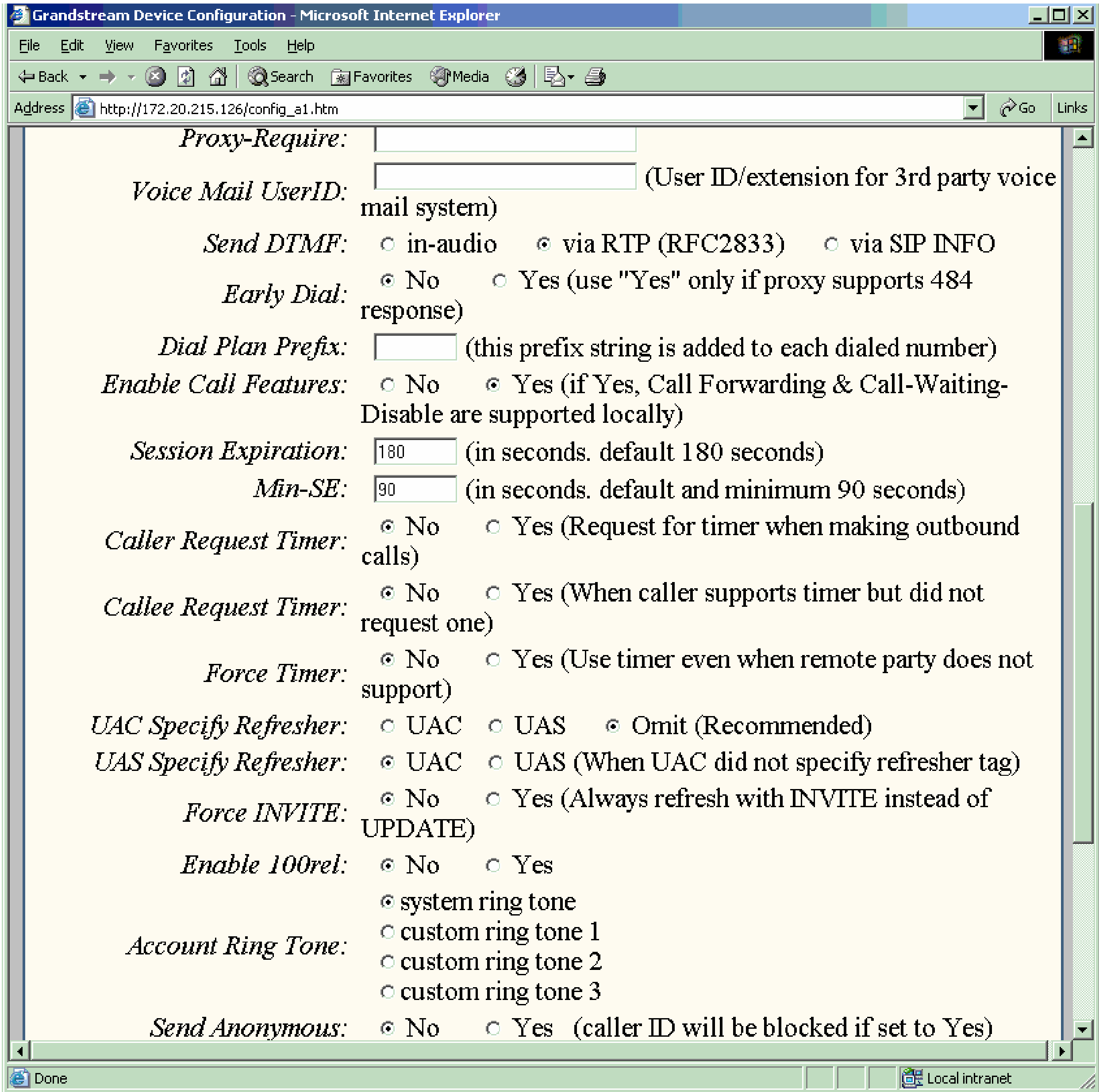

Grandstream Device Configuration - Microsoft Internet Explorer
File Edit View Favorites Tools Help
Back Search Favorites Media
Address http://172.20.215.126/config2.htm
Go Links
Multi Purpose Key 4: Key Mode: Speed Dial Account: Account 1
Name: UserID:
Multi Purpose Key 5: Key Mode: Speed Dial Account: Account 1
Name: UserID:
Multi Purpose Key 6: Key Mode: Speed Dial Account: Account 1
Name: UserID:
Multi Purpose Key 7: Key Mode: Speed Dial Account: Account 1
Name: UserID:
Time Zone: GMT-8:00 (US Pacific Time, Los Angeles)
Daylight Savings Time: No Yes (if set to Yes, display time will be 1 hour ahead of normal time)
LCD Backlight Always On: No Yes
Time Display Format: 12 HOUR 24 HOUR
Date Display Format: Year-Month-Day Month-Day-Year Day-Month-Year
Display Clock instead of Date: No Yes
System Device Mode
Device Mode: Switch (default) NAT/Router
Done
Local intranet

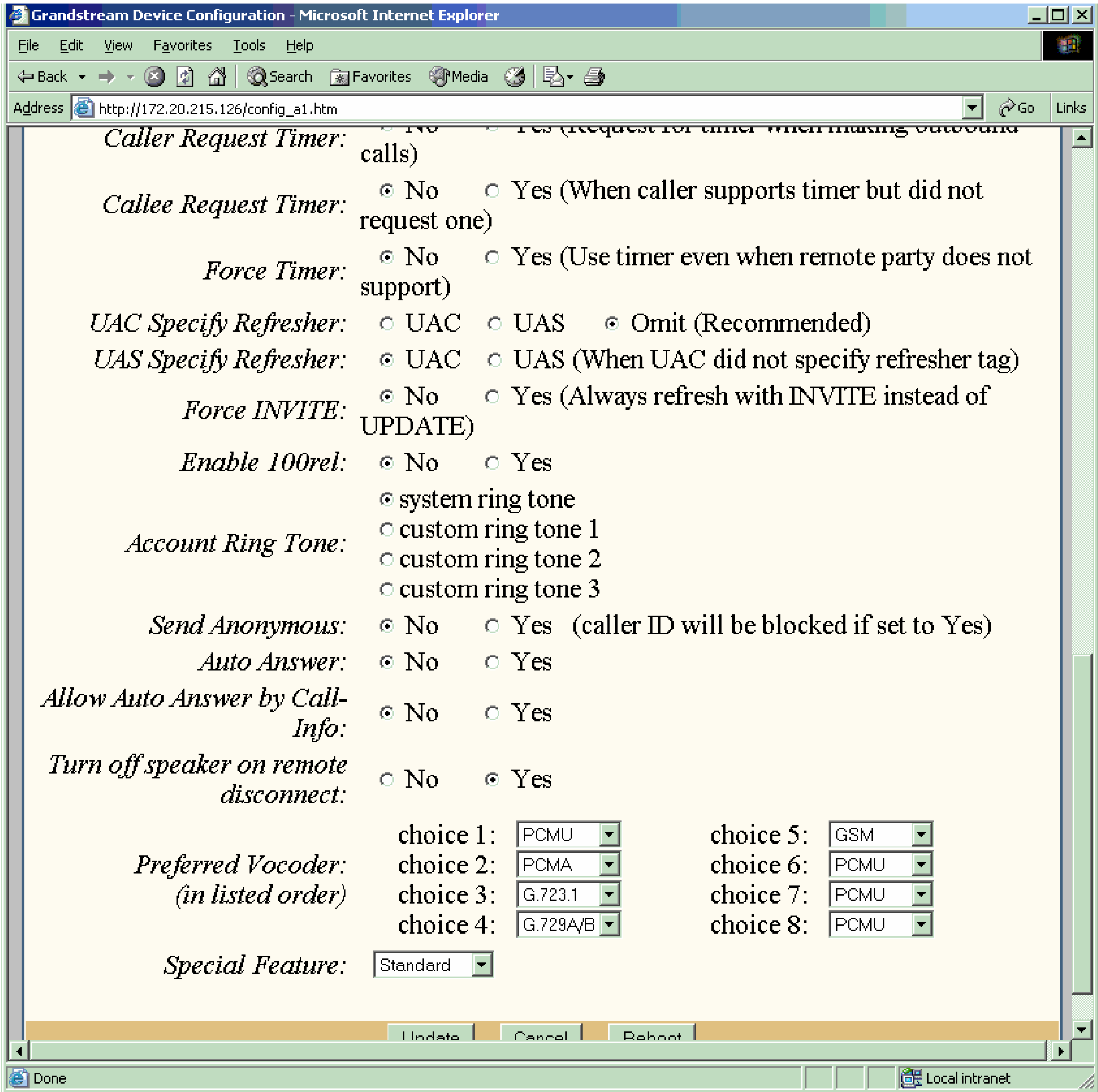

Grandstream Device Configuration - Microsoft Internet Explorer
File Edit View Favorites Tools Help
Back Search Favorites Media
Address http://172.20.215.126/config2.htm
Go Links
System Device Mode
Device Mode: Switch (default) NAT/Router
NAT/Router Configuration
WAN side http access: No Yes (WAN side access to http server will be rejected if set to No)
Reply to ICMP on WAN port: No Yes (Unit will not respond to PING from WAN side if set to No)
Cloned WAN MAC Addr: (in hex format)
LAN Subnet Mask: (default is 255.255.255.0)
LAN DHCP Base IP: (base IP for the LAN port, default is 192.168.2.1)
DHCP IP Lease Time: 120 (in units of hours, default is 120 hours or 5 days)
DMZ IP:
WAN port LAN IP LAN port Protocol
Port Forwarding:
0 0 UDP Only
0 0 UDP Only
0 0 UDP Only
0 0 UDP Only
0 0 UDP Only
0 0 UDP Only
0 0 UDP Only
0 0 UDP Only
Update Cancel Reboot
Done
Local intranet

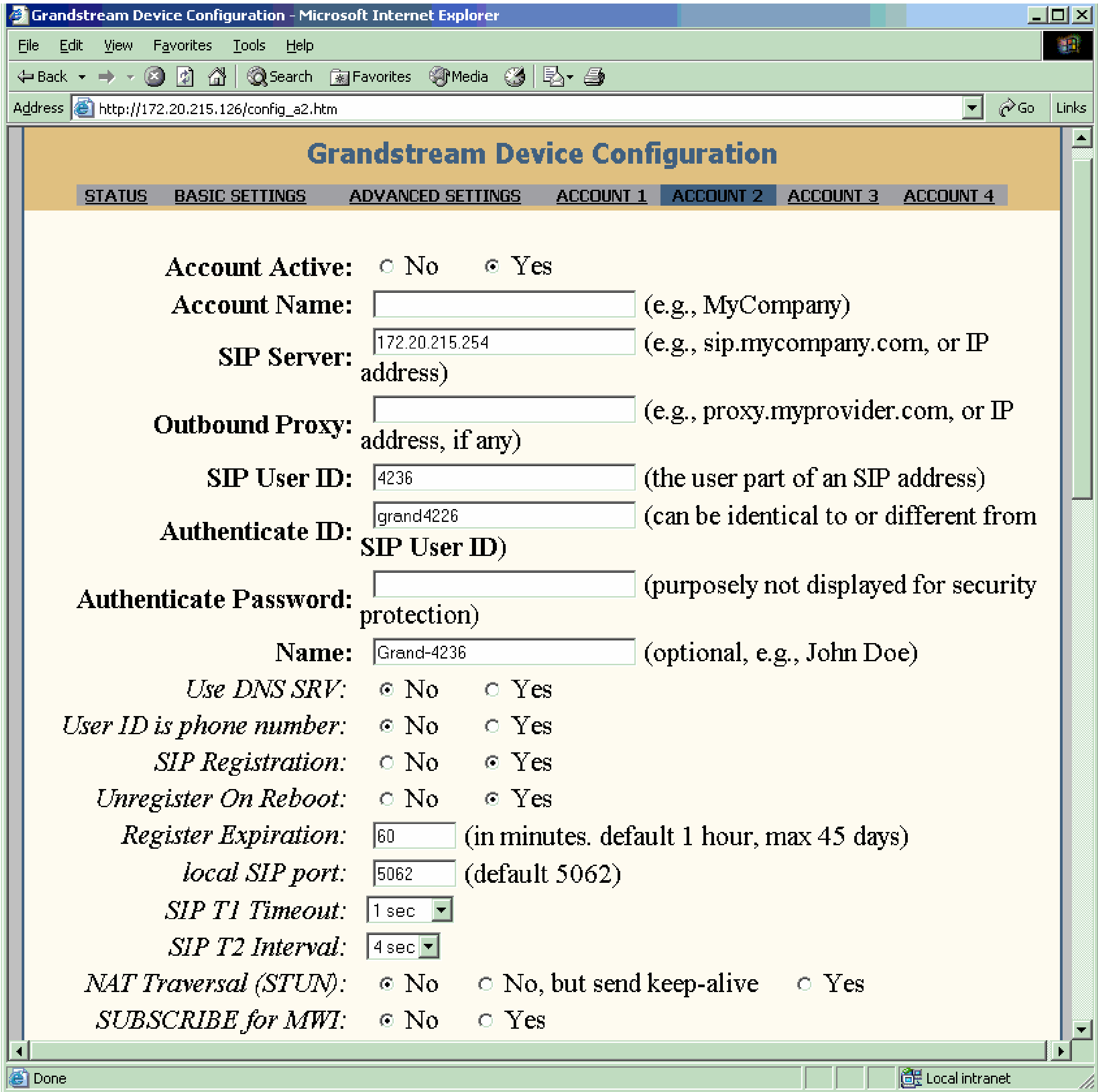

Grandstream Device Configuration - Microsoft Internet Explorer
File Edit View Favorites Tools Help
Back Search Favorites Media
Address http://172.20.215.126/config.htm
Go Links
Grandstream Device Configuration
STATUS BASIC SETTINGS ADVANCED SETTINGS ACCOUNT 1 ACCOUNT 2 ACCOUNT 3 ACCOUNT 4
Admin Password: (purposely not displayed for security protection)
Silence Suppression: No Yes
Voice Frames per TX: 2 (up to 10/20/32/64 for G711/G726/G723/other codecs respectively)
Layer 3 QoS: 48 (Diff-Serv or Precedence value)
Layer 2 QoS: 802.1Q/VLAN Tag 0 802.1p priority value 0 (0-7)
No Key Entry Timeout: 4 (in seconds, default is 4 seconds)
Use # as Dial Key: No Yes (if set to Yes, "#" will function as the "(Re-)Dial" key)
local RTP port: 5004 (1024-65535, default 5004)
Use random port: No Yes
keep-alive interval: 60 (in seconds, default 20 seconds)
Use NAT IP (if specified, this will be used in SIP/SDP message)
STUN server: (URI or IP:port)
Firmware Upgrade and Provisioning: Upgrade Via TFTP HTTP
Firmware Server Path: 172.20.2.181
Config Server Path: 172.20.2.181
Firmware File Prefix: Firmware File Postfix:
Config File Prefix: Config File Postfix:
Done
Local intranet

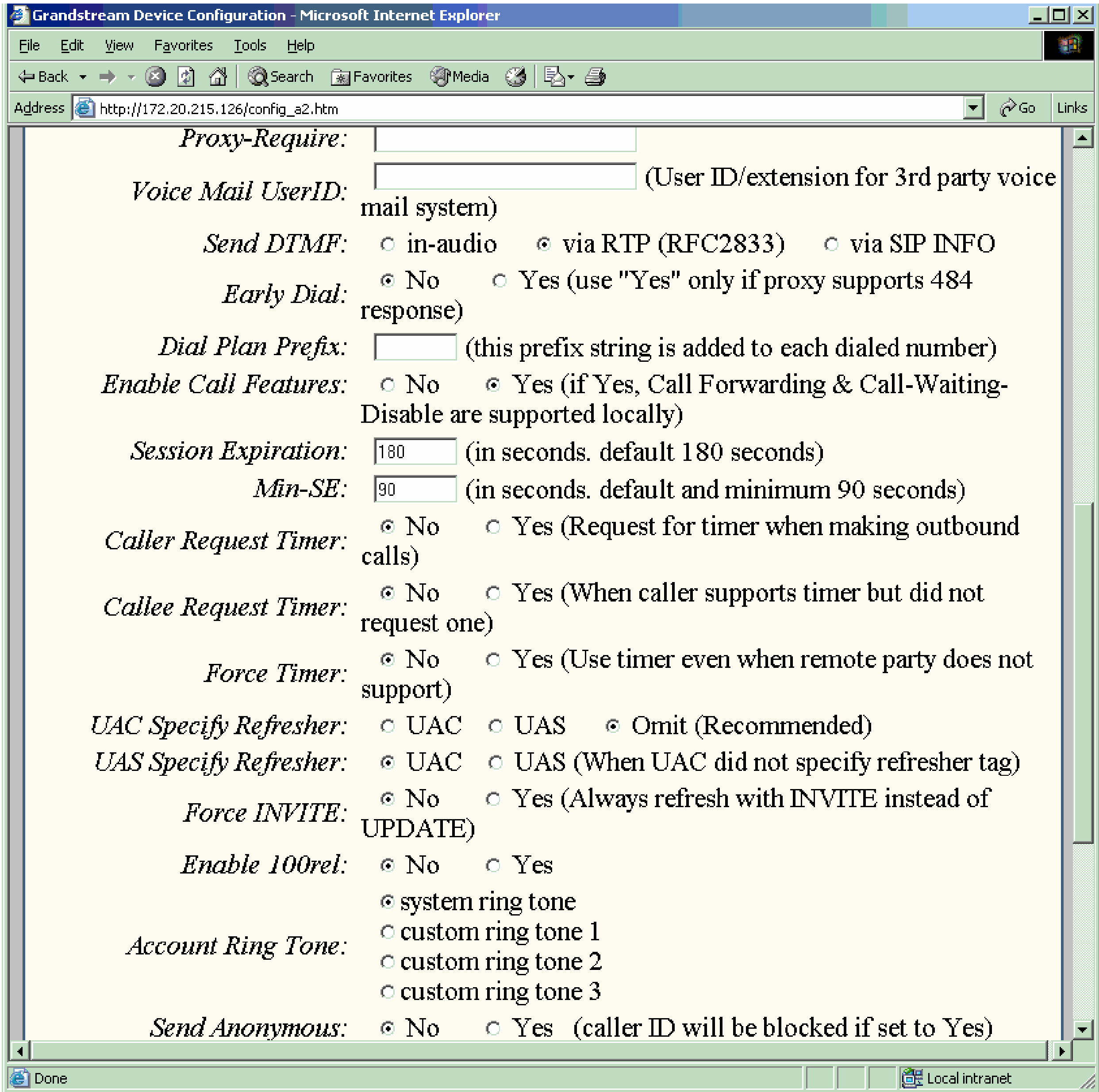

Grandstream Device Configuration - Microsoft Internet Explorer
File Edit View Favorites Tools Help
Back Search Favorites Media
Address http://172.20.215.126/config.htm
Go Links
Postfix:
Config File Prefix: Config File Postfix:
Automatic Upgrade:
No Yes, check for upgrade every 10080 minutes (default 7 days)
Always Check for New Firmware
Check New Firmware only when F/W pre/suffix changes
Authenticate Conf File: No Yes (cfg file would be authenticated before acceptance if set to Yes)
DTMF Payload Type: 101
Syslog Server:
Syslog Level: NONE
NTP Server: 172.20.2.181 (URI or IP address)
Distinctive Ring Tone: Custom ring tone 1, used if incoming caller ID is
Custom ring tone 2, used if incoming caller ID is
Custom ring tone 3, used if incoming caller ID is
Disable Call-Waiting: No Yes
Lock keypad update: No Yes (configuration update via keypad is disabled if set to Yes)
Update Cancel Reboot
All Rights Reserved Grandstream Networks, Inc. 2004, 2005
Done
Local intranet

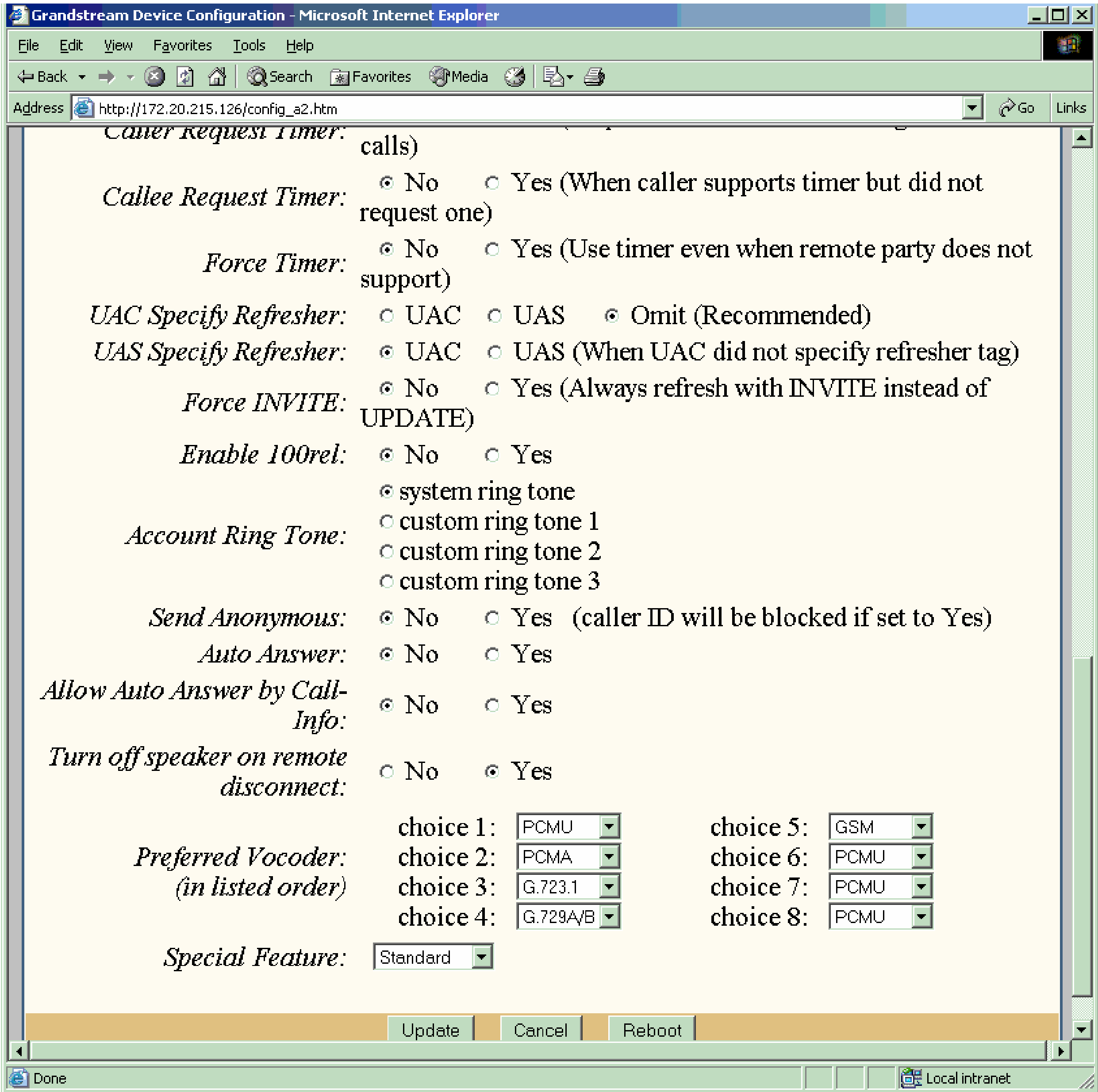

Grandstream Device Configuration - Microsoft Internet Explorer
File Edit View Favorites Tools Help
Back Search Favorites Media
Address http://172.20.215.126/config_a1.htm
Go Links
Grandstream Device Configuration
STATUS BASIC SETTINGS ADVANCED SETTINGS ACCOUNT 1 ACCOUNT 2 ACCOUNT 3 ACCOUNT 4
Account Active: No Yes
Account Name: (e.g., MyCompany)
SIP Server: 172.20.215.254 (e.g., sip.mycompany.com, or IP address)
Outbound Proxy: (e.g., proxy.myprovider.com, or IP address, if any)
SIP User ID: 4226 (the user part of an SIP address)
Authenticate ID: grand4226 (can be identical to or different from SIP User ID)
Authenticate Password: (purposely not displayed for security protection)
Name: Grand-4226 (optional, e.g., John Doe)
Use DNS SRV: No Yes
User ID is phone number: No Yes
SIP Registration: No Yes
Unregister On Reboot: No Yes
Register Expiration: 60 (in minutes. default 1 hour, max 45 days)
local SIP port: 5060 (default 5060)
SIP T1 Timeout: 1 sec
SIP T2 Interval: 4 sec
NAT Traversal (STUN): No No, but send keep-alive Yes
SUBSCRIBE for MWI: No Yes
Done
Local intranet


Grandstream Device Configuration - Microsoft Internet Explorer
File Edit View Favorites Tools Help
Back Search Favorites Media
Address http://172.20.215.126/config_a1.htm
Go Links
Proxy-Require:
Voice Mail UserID: (User ID/extension for 3rd party voice mail system)
Send DTMF: in-audio via RTP (RFC2833) via SIP INFO
Early Dial: No Yes (use "Yes" only if proxy supports 484 response)
Dial Plan Prefix: (this prefix string is added to each dialed number)
Enable Call Features: No Yes (if Yes, Call Forwarding & Call-Waiting-Disable are supported locally)
Session Expiration: 180 (in seconds. default 180 seconds)
Min-SE: 90 (in seconds. default and minimum 90 seconds)
Caller Request Timer: No Yes (Request for timer when making outbound calls)
Callee Request Timer: No Yes (When caller supports timer but did not request one)
Force Timer: No Yes (Use timer even when remote party does not support)
UAC Specify Refresher: UAC UAS Omit (Recommended)
UAS Specify Refresher: UAC UAS (When UAC did not specify refresher tag)
Force INVITE: No Yes (Always refresh with INVITE instead of UPDATE)
Enable 100rel: No Yes
Account Ring Tone: system ring tone
custom ring tone 1
custom ring tone 2
custom ring tone 3
Send Anonymous: No Yes (caller ID will be blocked if set to Yes)
Done
Local intranet


Grandstream Device Configuration - Microsoft Internet Explorer
File Edit View Favorites Tools Help
Back Search Favorites Media
Address http://172.20.215.126/config_a1.htm
Go Links
Caller Request Timer: calls)
Callee Request Timer: No Yes (When caller supports timer but did not request one)
Force Timer: No Yes (Use timer even when remote party does not support)
UAC Specify Refresher: UAC UAS Omit (Recommended)
UAS Specify Refresher: UAC UAS (When UAC did not specify refresher tag)
Force INVITE: No Yes (Always refresh with INVITE instead of UPDATE)
Enable 100rel: No Yes
Account Ring Tone: system ring tone custom ring tone 1 custom ring tone 2 custom ring tone 3
Send Anonymous: No Yes (caller ID will be blocked if set to Yes)
Auto Answer: No Yes
Allow Auto Answer by Call-Info: No Yes
Turn off speaker on remote disconnect: No Yes
Preferred Vocoder: (in listed order)
choice 1: PCMU choice 5: GSM
choice 2: PCMA choice 6: PCMU
choice 3: G.723.1 choice 7: PCMU
choice 4: G.729A/B choice 8: PCMU
Special Feature: Standard
Done
Local intranet


Grandstream Device Configuration - Microsoft Internet Explorer
File Edit View Favorites Tools Help
Back Search Favorites Media
Address http://172.20.215.126/config_a2.htm
Go Links
Grandstream Device Configuration
STATUS BASIC SETTINGS ADVANCED SETTINGS ACCOUNT 1 ACCOUNT 2 ACCOUNT 3 ACCOUNT 4
Account Active: No Yes
Account Name: (e.g., MyCompany)
SIP Server: 172.20.215.254 (e.g., sip.mycompany.com, or IP address)
Outbound Proxy: (e.g., proxy.myprovider.com, or IP address, if any)
SIP User ID: 4236 (the user part of an SIP address)
Authenticate ID: grand4226 (can be identical to or different from SIP User ID)
Authenticate Password: (purposely not displayed for security protection)
Name: Grand-4236 (optional, e.g., John Doe)
Use DNS SRV: No Yes
User ID is phone number: No Yes
SIP Registration: No Yes
Unregister On Reboot: No Yes
Register Expiration: 60 (in minutes. default 1 hour, max 45 days)
local SIP port: 5062 (default 5062)
SIP T1 Timeout: 1 sec
SIP T2 Interval: 4 sec
NAT Traversal (STUN): No No, but send keep-alive Yes
SUBSCRIBE for MWI: No Yes
Done
Local intranet


Grandstream Device Configuration - Microsoft Internet Explorer
File Edit View Favorites Tools Help
Back Search Favorites Media
Address http://172.20.215.126/config_a2.htm
Go Links
Proxy-Require:
Voice Mail UserID: (User ID/extension for 3rd party voice mail system)
Send DTMF: in-audio via RTP (RFC2833) via SIP INFO
Early Dial: No Yes (use "Yes" only if proxy supports 484 response)
Dial Plan Prefix: (this prefix string is added to each dialed number)
Enable Call Features: No Yes (if Yes, Call Forwarding & Call-Waiting-Disable are supported locally)
Session Expiration: 180 (in seconds. default 180 seconds)
Min-SE: 90 (in seconds. default and minimum 90 seconds)
Caller Request Timer: No Yes (Request for timer when making outbound calls)
Callee Request Timer: No Yes (When caller supports timer but did not request one)
Force Timer: No Yes (Use timer even when remote party does not support)
UAC Specify Refresher: UAC UAS Omit (Recommended)
UAS Specify Refresher: UAC UAS (When UAC did not specify refresher tag)
Force INVITE: No Yes (Always refresh with INVITE instead of UPDATE)
Enable 100rel: No Yes
Account Ring Tone: system ring tone
custom ring tone 1
custom ring tone 2
custom ring tone 3
Send Anonymous: No Yes (caller ID will be blocked if set to Yes)
Done
Local intranet


Grandstream Device Configuration - Microsoft Internet Explorer
File Edit View Favorites Tools Help
Back Search Favorites Media
Address http://172.20.215.126/config_a2.htm
Go Links
calls)
Callee Request Timer: No Yes (When caller supports timer but did not request one)
Force Timer: No Yes (Use timer even when remote party does not support)
UAC Specify Refresher: UAC UAS Omit (Recommended)
UAS Specify Refresher: UAC UAS (When UAC did not specify refresher tag)
Force INVITE: No Yes (Always refresh with INVITE instead of UPDATE)
Enable 100rel: No Yes
Account Ring Tone: system ring tone custom ring tone 1 custom ring tone 2 custom ring tone 3
Send Anonymous: No Yes (caller ID will be blocked if set to Yes)
Auto Answer: No Yes
Allow Auto Answer by Call-Info: No Yes
Turn off speaker on remote disconnect: No Yes
Preferred Vocoder: (in listed order)
choice 1: PCMU
choice 2: PCMA
choice 3: G.723.1
choice 4: G.729A/B
choice 5: GSM
choice 6: PCMU
choice 7: PCMU
choice 8: PCMU
Special Feature: Standard
Update Cancel Reboot
Done
Local intranet