



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.

TABLE OF CONTENTS

1	INTRODUCTION	6
1.1	SCOPE	6
1.2	BACKGROUND FOR COMPATIBILITY/INTEROPERABILITY TESTS	7
1.3	REFERENCES	7
1.4	TERMINOLOGY.....	8
1.4.1	General.....	8
2	SUMMARY TEST REPORT	9
2.1	NETWORK DIAGRAM	9
2.2	TEST RESULT MATRIX	10
2.3	TEST RESULT SUMMARY	11
3	SIP PHONE REGISTRATION	12
3.1	TEST CASES FOR SIP PHONE REGISTRATION	12
3.1.1	SIP phone Registration.....	12
3.1.2	SIP phone Unregistration.....	13
3.1.3	SIP phone Registration – Multiple Lines.....	14
3.1.4	SIP phone support of HTTP Digest.....	14
4	BASIC CALL.....	15
4.1	TEST CASES FOR BASIC CALL	15
4.1.1	Station-to-Station call SIP call.....	15
4.1.2	Incoming and Outgoing Voice Calls with Line Identification	16
4.1.2.1	Voice call from vendor SIP phone to Cisco SIP Phone – number presentation	16
4.1.2.2	Voice call from Cisco SIP Phone to vendor SIP phone – number presentation	16
4.1.2.3	Voice call from vendor SIP phone to Cisco IP Phone (SCCP) – number presentation	17
4.1.2.4	Voice call from Cisco IP Phone (SCCP) to vendor SIP phone– number presentation	18
4.1.3	Unsuccessful Call Termination and Call Clearing	18
4.1.3.1	Outgoing Call vendor SIP to Cisco SIP - No Answer.....	18
4.1.3.2	Outgoing Call Cisco SIP to vendor SIP - No Answer.....	19
4.1.3.3	Outgoing Call vendor SIP to Cisco SIP - Busy Station.....	19
4.1.3.4	Outgoing Call Cisco SIP to vendor SIP - Busy Station.....	20
4.1.3.5	Outgoing Call vendor SIP to Cisco SCCP - No Answer.....	20
4.1.3.6	Outgoing Call Cisco SCCP to vendor SIP - No Answer.....	21
4.1.3.7	Outgoing Call vendor SIP to Cisco SCCP - Busy Station.....	21
4.1.3.8	Outgoing Call Cisco SCCP to vendor SIP - Busy Station.....	22
5	CALLING/CONNECTED NAME IDENTIFICATION	22
5.1	DESCRIPTION	22
5.2	TEST CASES	22
5.2.1	Vendor SIP to Cisco SIP Voice call with Calling/Connected Name Identification	22
5.2.2	Cisco SIP to vendor SIP Voice call with Calling/Connected Name Identification.....	23
5.2.3	Vendor SIP to Cisco SCCP Voice call with Calling/Connected Name Identification	24
5.2.4	Cisco SCCP to vendor SIP Voice call with Calling/Connected Name Identification	24
6	ALERTING NAME IDENTIFICATION.....	25
6.1	DESCRIPTION	25
6.2	TEST CASES	25
6.2.1	Vendor SIP to Cisco SIP Voice call with Alerting Name enabled.....	25
6.2.2	Vendor SIP to Cisco SCCP Voice call with Alerting Name enabled.....	26
6.2.3	Cisco SCCP to vendor SIP Voice call with Alerting Name enabled.....	27
6.2.4	Vendor SIP to Cisco SIP Voice call with Alerting Name restricted	27
6.2.5	Cisco SIP to vendor SIP Voice call with Alerting Name restricted.....	28

6.2.6	Cisco SCCP to vendor SIP Voice call with Alerting Name restricted.....	28
7	HOLD & RESUME	29
7.1	TEST CASES FOR HOLD & RESUME	29
7.1.1	Vendor SIP to Cisco SIP Voice call – originating end initiates the Hold	29
7.1.2	Vendor SIP to Cisco SIP Voice call – terminating end initiates the Hold.....	29
7.1.3	Vendor SIP to Cisco SCCP Voice call – originating end initiates the Hold	30
7.1.4	Vendor SIP to Cisco SCCP Voice call – terminating end initiates the Hold.....	31
7.1.5	Cisco SCCP to vendor SIP Voice call – originating end initiates the Hold	32
7.1.6	Cisco SCCP to vendor SIP Voice call – terminating end initiates the Hold	32
8	CALL TRANSFER.....	33
8.1	DESCRIPTION	33
8.1.1	Attended.....	33
8.1.2	Early Attended.....	33
8.1.3	Blind.....	34
8.2	CALL TRANSFER TEST CASES	34
8.2.1	Vendor SIP to vendor SIP Attended Call Transfer completes – secondary call answered.....	34
8.2.2	Vendor SIP to Cisco SCCP Attended Call Transfer completes – secondary call answered.....	35
8.2.3	Vendor SIP to Cisco SIP Attended Call Transfer completes – secondary call answered.....	35
8.2.4	Cisco SCCP to vendor SIP Attended Call Transfer completes – secondary call answered	36
8.2.5	Vendor SIP to vendor SIP Early Attended Call Transfer completes – secondary call answered.....	37
8.2.6	Vendor SIP to Cisco SIP Early Attended Call Transfer completes – secondary call answered.....	37
8.2.7	Vendor SIP to Cisco SCCP Early Attended Call Transfer completes – secondary call answered.....	38
8.2.8	Cisco SCCP to vendor SIP Early Attended Call Transfer completes – secondary call answered	39
8.2.9	Vendor SIP to vendor SIP Blind Call Transfer completes – call answered	39
8.2.10	Vendor SIP to Cisco SCCP Blind Call Transfer completes – call answered	40
8.2.11	Cisco SCCP to Vendor SIP Blind Call Transfer completes – call answered	41
9	CALL DIVERSION (CFU-CALL FORWARDING UNCONDITIONAL), (CFB-CALL FORWARDING BUSY), (CFNA-CALL FORWARDING NO ANSWER)	41
9.1	DESCRIPTION	41
9.2	TEST CONFIGURATION FOR DIVERSION	42
9.3	TEST CASES	42
9.3.1	Vendor SIP to Cisco SIP Call Forwarding	42
9.3.2	Vendor SIP to Vendor SIP Call Forwarding.....	43
9.3.3	Vendor SIP to Cisco SCCP Call Forwarding.....	44
10	CALL CONFERENCE	46
10.1	DESCRIPTION	46
10.2	TEST CONFIGURATION FOR CONFERENCE.....	46
10.3	CALL CONFERENCE TEST CASES	46
10.3.1	Vendor SIP to Vendor SIP Call Conference.....	46
10.3.2	Vendor SIP to Cisco SIP Call Conference	47
10.3.3	Vendor SIP to Cisco SCCP Call Conference	48
10.3.4	Cisco SCCP to Vendor SIP Call Conference	48
11	VOICEMAIL ACCESS AND MWI SUPPORT.	49
11.1	DESCRIPTION	49
11.2	TEST CONFIGURATION FOR VOICEMAIL AND MWI SUPPORT.....	49
11.3	TEST CASES	49
11.3.1	Generic voicemail access/ DTMF tone outpulsing to Cisco Unity ports.....	49
11.3.2	Message Waiting Indication – MWI on	50
11.3.3	Message Waiting Indication – MWI off.....	50

12	PRODUCT CONFIGURATIONS.....	52
12.1	CISCO CCM CONFIGURATION	52
12.1.1	CCM Version.....	52
12.1.2	Cisco 7960 Phone Configurations.....	53
12.1.3	Cisco 7961 SIP Phone Configurations.....	60
12.1.4	Route Pattern for Unity Access	67
12.1.5	SIP Trunk for Unity Access Configuration.....	69
12.1.6	Default SIP Profile Configuration.....	71
12.1.7	Generic SIP Phone Security Profile Configuration.....	73
12.1.8	User ID List for 3 rd Party Phones.....	74
12.1.9	End User Configuration for 3 rd Party Phone	75
12.1.10	Partitions Configuration	77
12.1.11	Incoming Trunk CSS Configuration	78
12.1.12	Phones CSS Configuration.....	79
12.1.13	Media Resource Group	80
12.1.14	Media Resource Group List.....	81
12.1.15	Voicemail Profile Configuration	82
12.1.16	Voicemail Pilot.....	83
12.1.17	Voicemail Ports.....	84
12.1.18	MWI Configuration	87
12.1.19	Translation Pattern for Voicemail Access.....	89
12.2	3 RD PARTY SIP PHONES CCM CONFIGURATION	90
12.2.1	Grandstream Budgetone 100.....	90
12.2.2	Grandstream GXP-2000.....	95
12.3	GRANDSTREAM BUDGETONE 100 SIP PHONE WEBPAGE CONFIGURATION	100
12.4	GRANDSTREAM GXP-2000 SIP PHONE WEBPAGE CONFIGURATION.....	106

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 CallManager 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

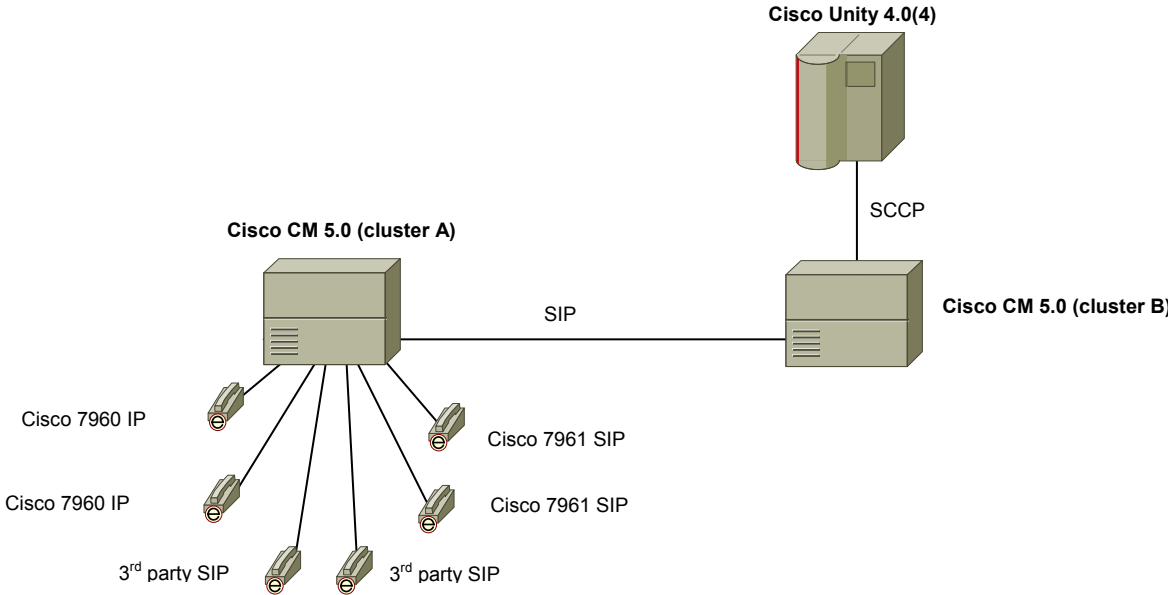
1.4.1 General

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 Grandstream 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 3 rd Party SIP phone initiated calls, when the call is put on hold either from 3 rd 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 config 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 Grandstream 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 Grandstream 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 Grandstream 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 (transferring) 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 (transferring) 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 (transferring) 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 (transferring) 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 vendor 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 vendor 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 updated 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

Cisco CallManager Console - Microsoft Internet Explorer provided by Cisco Systems, Inc.

File Edit View Favorites Tools Help

Back Forward Stop Refresh Home Search Favorites Media

Address <https://172.25.67.126:8443/ccmadmin/showHome.do> 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

 **Cisco CallManager Administration**

System version: 5.0.1.51-410
Administration version: 1.1.0.0-1

Copyright © 1999 - 2005 Cisco Systems, Inc.
All rights reserved.

This product contains cryptographic features and is subject to United States and local country laws governing import, export, transfer and use. Delivery of Cisco cryptographic products does not imply third-party authority to import, export, distribute or use encryption. Importers, exporters, distributors and users are responsible for compliance with U.S. and local country laws. By using this product you agree to comply with applicable laws and regulations. If you are unable to comply with U.S. and local laws, return this product immediately.

A summary of U.S. laws governing Cisco cryptographic products may be found at: <http://www.cisco.com/www/export/crypto/tool/starq.html>.
If you require further assistance please contact us by sending email to export@cisco.com.

Done Internet

12.1.2 Cisco 7960 Phone Configurations

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

Address: <https://172.25.67.126:8443/ccmadmin/phoneEdit.do?key=aad31f4b-f74c-4749-92b7-fae9e8b1c637>

Association Information

- Modify Button Items
- 1 [Line \[1\] - 4201 in Phones](#)
- 2 [Line \[2\] - Add a new DN](#)
- 3 [Add a new SD](#)
- 4 [Add a new SD](#)
- 5 [Add a new SD](#)
- 6 [Add a new SD](#)
- Unassigned Associated Items -----
- 7 [Add a new SD](#)
- 8 [Add a new SURL](#)
- 9 [Add a new BLF SD](#)
- 10 Privacy
- 11 None

Phone Type

Product Type: Cisco 7960
Device Protocol: SCCP

Device Information

Registration: Registered with Cisco CallManager CM-MERCURY

IP Address: [172.20.215.101](#)

MAC Address*: 000A8AA20CD6

Description: 4201 SCCP

Device Pool*: Default

Phone Button Template*: Standard 7960 SCCP

Softkey Template: Standard User CallBack

Common Phone Profile*: Standard Common Phone Profile

Calling Search Space: < None >

AAR Calling Search Space: < None >

Media Resource Group List: MRGL-CM-MERCURY

User Hold Audio Source: 1-SampleAudioSource

Network Hold Audio Source: < None >

Location*: Hub_None

User Locale: < None >

Network Locale: < None >

Built In Bridge*: Default

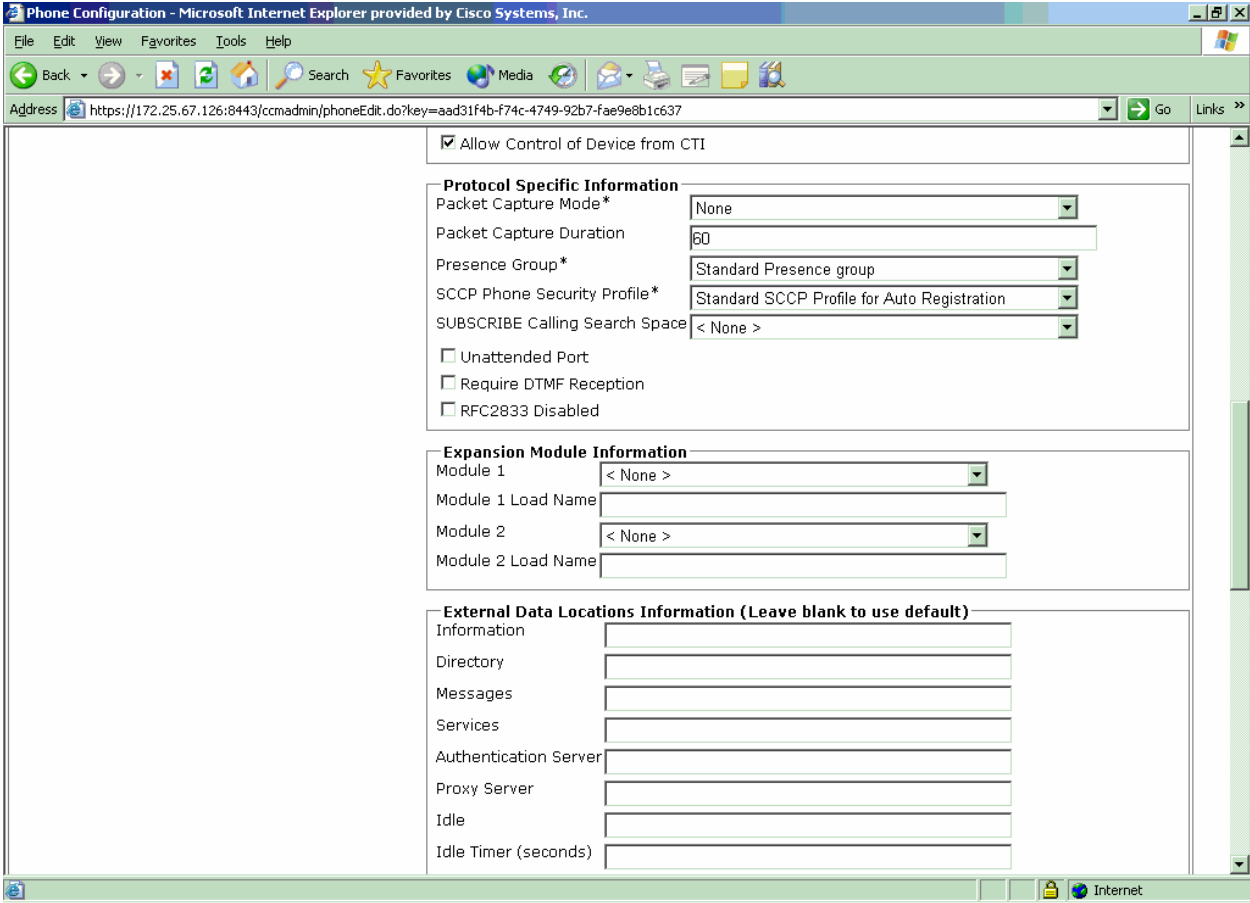
Privacy*: Default

Owner User ID: < None >

Phone Load Name:

Retry Video Call as Audio
 Ignore Presentation Indicators (internal calls only)





The screenshot shows a Microsoft Internet Explorer browser window titled "Phone Configuration - Microsoft Internet Explorer provided by Cisco Systems, Inc.". The address bar contains the URL: <https://172.25.67.126:8443/ccmadmin/phoneEdit.do?key=aad31f4b-f74c-4749-92b7-fae9e8b1c637>. The page content is organized into several sections:

- Extension Information:** Includes a checkbox for "Enable Extension Mobility", a "Log Out Profile" dropdown menu (set to "-- Not Selected --"), and fields for "Login in User ID", "Log in Time", and "Log out Time", all currently set to "< None >".
- Certification Authority Proxy Function (CAPF) Information:** Features a "Certificate Operation*" dropdown (set to "No Pending Operation"), an "Authentication String" text input, a "Generate String" button, and "Operation Completes By" fields (set to 2006, 2, 18, 12). The status is "Certificate Operation Status: None".
- MLPP Information:** Contains dropdown menus for "MLPP Domain" (set to "< None >"), "MLPP Indication*" (set to "Default"), and "MLPP Preemption*" (set to "Default").
- Secure Shell Information:** Includes text input fields for "Secure Shell User" and "Secure Shell Password".
- Product Specific Configuration:** Contains checkboxes for "Disable Speakerphone" and "Disable Speakerphone and Headset", and dropdown menus for "PC Port *" (set to "Enabled"), "Settings Access *" (set to "Enabled"), and "Gratuitous ARP *" (set to "Enabled").

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

File Edit View Favorites Tools Help

Back Forward Stop Refresh Home 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

i *- indicates required item.
i - 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

Address <https://172.25.67.126:8443/ccmadmin/directoryNumberEdit.do?key=cce65846-4a77-45d3-a5c2-9304a3627661&mapkey=deb62f2c-eb79-4b25-b6dd-bb855ef005b3&device> 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	

Call Forward and Call Pickup Settings

Done Internet



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

File Edit View Favorites Tools Help

Back Forward Stop Refresh Home 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

Call Forward and Call Pickup Settings

	Voice Mail Destination	Calling Search Space
Forward All	<input type="checkbox"/> or <input type="text"/>	Phones
Secondary Calling Search Space for Forward All		< None > Find
Forward Busy Internal	<input type="checkbox"/> or <input type="text"/>	Phones
Forward Busy External	<input type="checkbox"/> or <input type="text"/>	Phones
Forward No Answer Internal	<input type="checkbox"/> or <input type="text"/>	Phones
Forward No Answer External	<input type="checkbox"/> or <input type="text"/>	Phones
Forward No Coverage Internal	<input type="checkbox"/> or <input type="text"/>	< None >
Forward No Coverage External	<input type="checkbox"/> or <input type="text"/>	< None >
Forward on CTI Failure	<input type="checkbox"/> or <input type="text"/>	< None >
No Answer Ring Duration (seconds)	<input type="text"/>	
Call Pickup Group	< None >	

MLPP Alternate Party Settings

Target (Destination)	<input type="text"/>
MLPP Calling Search Space	< None >
MLPP No Answer Ring Duration (seconds)	<input type="text"/>

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.

Address: https://172.25.67.126:8443/ccmadmin/directoryNumberEdit.do?key=cc65846-4a77-45d3-a5c2-9304a3627661&mapkey=deb62f2c-eb79-4b25-b6dd-bb855ef005b3&devic

Line 1 on Device SEP000A8AA20CD6

Display (Internal Caller ID) 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)

Line Text Label

ASCII Line Text Label

External Phone Number Mask

Message Waiting Lamp Policy*

Ring Setting (Phone Idle)*

Ring Setting (Phone Active) 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*

Busy Trigger* (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.



12.1.3 Cisco 7961 SIP Phone Configurations

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

Address: <https://172.25.67.126:8443/ccmadmin/phoneEdit.do?key=f96fff94-deeb-8a18-ec9-1237ef157561>

Association Information

Modify Button Items

- 1 [Line \[1\] - 4208 in Phones](#)
- 2 [Line \[2\] - Add a new DN](#)
- 3 [Add a new SD](#)
- 4 [Add a new SD](#)
- 5 [Add a new SD](#)
- 6 [Add a new SD](#)
- 7 ----- Unassigned Associated Items -----
- 8 [Add a new SD](#)
- 9 [Add a new SURL](#)
- 10 [Add a new BLF SD](#)
- 10 Privacy
- 11 None

Phone Type
Product Type: Cisco 7961
Device Protocol: SIP

Device Information

Registration: Registered with Cisco CallManager CM-MERCURY

IP Address: [172.20.215.106](#)

MAC Address*: 00152B8F3967

Description: 4208 SIP

Device Pool*: Default

Phone Button Template*: Standard 7961 SIP

Softkey Template: Standard User CallBack

Common Phone Profile*: Standard Common Phone Profile

Calling Search Space: < None >

AAR Calling Search Space: < None >

Media Resource Group List: MRGL-CM-MERCURY

User Hold Audio Source: 1-SampleAudioSource

Network Hold Audio Source: 1-SampleAudioSource

Location*: Hub_None

User Locale: English United States

Network Locale: < None >

Built In Bridge*: Default

Privacy*: Off

Owner User ID: < None >

Phone Load Name:

Ignore Presentation Indicators (internal calls only)

Allow Control of Device from CTI

The screenshot shows a web browser window titled "Phone Configuration - Microsoft Internet Explorer provided by Cisco Systems, Inc.". The address bar contains the URL: <https://172.25.67.126:8443/ccmadmin/phoneEdit.do?key=f96fff94-deeb-8a18-ec9-1237ef157561>. The main content area is divided into three sections:

- Protocol Specific Information**: Contains several dropdown menus and checkboxes. The dropdowns are: 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), and Digest User (< None >). There are three checkboxes: Media Termination Point Required, Unattended Port, and Require DTMF Reception.
- External Data Locations Information (Leave blank to use default)**: Contains several empty text input fields for: Information, Directory, Messages, Services, Authentication Server, Proxy Server, Idle, and Idle Timer (seconds).
- Extension Information**: Contains one checkbox: Enable Extension Mobility.

The browser's status bar at the bottom shows "Done" on the left and "Internet" on the right.

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

Address: <https://172.25.67.126:8443/ccmadmin/phoneEdit.do?key=f96fff94-deeb-8a18-ec9-1237ef157561>

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:

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 *: Disabled



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

Address: https://172.25.67.126:8443/ccmadmin/phoneEdit.do?key=f96fff94-deeb-8a18-ec9-1237ef157561

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.



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

File Edit View Favorites Tools Help

Back Forward Stop Refresh Home 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

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	<input type="checkbox"/> or <input type="text"/>	Phones
Secondary Calling Search Space for Forward All		< None > Find
Forward Busy Internal	<input type="checkbox"/> or <input type="text"/>	Phones
Forward Busy External	<input type="checkbox"/> or <input type="text"/>	Phones
Forward No Answer Internal	<input type="checkbox"/> or <input type="text"/>	Phones
Forward No Answer External	<input type="checkbox"/> or <input type="text"/>	Phones
Forward No Coverage Internal	<input type="checkbox"/> or <input type="text"/>	< None >
Forward No Coverage External	<input type="checkbox"/> or <input type="text"/>	< None >
Forward on CTI Failure	<input type="checkbox"/> or <input type="text"/>	< None >
No Answer Ring Duration (seconds)	<input type="text"/>	
Call Pickup Group	< None >	

MLPP Alternate Party Settings

Target (Destination)	<input type="text"/>
MLPP Calling Search Space	< None >
MLPP No Answer Ring Duration (seconds)	<input type="text"/>

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.

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

Line 1 on Device SEP00152B8F3967

Display (Internal Caller ID) 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)

Line Text Label

ASCII Line Text Label

External Phone Number Mask

Message Waiting Lamp Policy*

Ring Setting (Phone Idle)*

Ring Setting (Phone Active) 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*

Busy Trigger* (Less than or equal to Max. Calls)

Forwarded Call Information Display on Device SEP00152B8F3967

- Caller Name
- Caller Number
- Redirected Number
- Dialed Number

Save Delete Copy Reset Add New

* - indicates required item.



12.1.4 Route Pattern for Unity Access

The screenshot shows a web browser window titled "Route Pattern Configuration - Microsoft Internet Explorer provided by Cisco Systems, Inc.". The address bar shows the URL: <https://172.25.67.126:8443/ccadmin/routePattern2Edit.do?key=11ca3433-67f2-2622-396b-7ccc3ecb3be7>. The main content area is divided into several sections:

- Pattern Definition:**
 - Route Pattern*: 6XXX
 - Route Partition: < None >
 - Description: to CM-VENUS
 - Numbering Plan: -- Not Selected --
 - Route Filter: < None >
 - MLPP Precedence*: Default
 - Gateway/Route List*: CM-VENUS-SIP (Edit)
 - Route Option: Route this pattern, Block this pattern (No Error)
 - Call Classification*: OffNet
 - Options: Allow Device Override, Provide Outside Dial Tone, Allow Overlap Sending, Urgent Priority, Require Forced Authorization Code
 - Authorization Level*: 0
 - Require Client Matter Code
- Calling Party Transformations:**
 - Use Calling Party's External Phone Number Mask
 - Calling Party Transform Mask: [Empty]
 - Prefix Digits (Outgoing Calls): [Empty]
 - Calling Line ID Presentation*: Default
 - Calling Name Presentation*: Default
- Connected Party Transformations:**
 - Connected Line ID Presentation*: Default
 - Connected Name Presentation*: Default
- Called Party Transformations:** [Section header, no visible content]



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

File Edit View Favorites Tools Help

Address <https://172.25.67.126:8443/ccmadmin/routePattern2Edit.do?key=11ca3433-67f2-2622-396b-7ccc3ecb3be7> Go Links >>

Authorization Level*
 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*
Calling Name Presentation*

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

Called Party Transformations
Discard Digits
Called Party Transform Mask
Prefix Digits (Outgoing Calls)

ISDN Network-Specific Facilities Information Element
Network Service Protocol
Carrier Identification Code
Network Service Service Parameter Name Service Parameter Value

*- indicates required item.

Done Internet



12.1.5 SIP Trunk for Unity Access Configuration

Device Information

Product: SIP Trunk
Device Protocol: SIP
Device Name*: CM-VENUS-SIP
Description: to CM-VENUS SIP Trunk
Device Pool*: Default
Call Classification*: Use System Default
Media Resource Group List: MRGL-CM-MERCURY
Location*: Hub_None
AAR Group: <None >
Packet Capture Mode*: None
Packet Capture Duration: 60

Media Termination Point Required
 Retry Video Call as Audio
 Transmit UTF-8 for Calling Party Name
 Unattended Port

Multilevel Precedence and Preemption (MLPP) Information

MLPP Domain: <None >

Call Routing Information

Inbound Calls

Significant Digits*: All
Connected Line ID Presentation*: Default
Connected Name Presentation*: Default
Calling Search Space: Incoming Trunk
AAR Calling Search Space: <None >
Prefix DN:



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

Address: https://172.25.67.126:8443/ccmadmin/trunkEdit.do?key=dce3ed9b-0046-4284-a752-a4766a21f2bf

Redirecting Diversion Header Delivery - Inbound

Outbound Calls

Calling Party Selection*

Calling Line ID Presentation*

Calling Name Presentation*

Caller ID DN

Caller Name

Redirecting Diversion Header Delivery - Outbound

SIP Information

Destination Address*

Destination Address is an SRV

Destination Port* Note: 0 indicates destination is SRV

MTP Preferred Originating Codec*

Presence Group*

SIP Trunk Security Profile*

Rerouting Calling Search Space

Out-Of-Dialog Refer Calling Search Space

SUBSCRIBE Calling Search Space

SIP Profile*

DTMF Signaling Method

Save Delete Reset Add New

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



12.1.6 Default SIP Profile Configuration

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



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

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

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.



12.1.7 Generic SIP Phone Security Profile Configuration

The screenshot shows the Cisco CallManager Administration web interface in Microsoft Internet Explorer. The browser title is "SIP Phone Security Profile Configuration - Microsoft Internet Explorer provided by Cisco Systems, Inc.". The address bar shows the URL: <https://172.25.67.126:8443/ccmadmin/sipPhoneSecurityProfileEdit.do?key=f0b00721-5395-49d2-9265-eb23227ceb35>. The page header includes "Cisco CallManager Administration" and "Logged in as: CCMAdministrator". A navigation menu contains: System, Call Routing, Media Resources, Voice Mail, Device, Application, User Management, Bulk Administration, and Help. A "Log Off" button is also present. The main content area is titled "SIP Phone Security Profile Configuration" and includes a "Related Links" section with a "Back To Find/List" button. Below this are several sections: "Status" (Status: Ready), "SIP Phone Security Profile Information" (Name: generic sip phones, Description, Nonce Validity Time: 600, Device Security Mode: Non Secure, Transport Type: UDP, Enable Digest Authentication checkbox), "SIP Phone Security Profile CAPF Information" (Authentication Mode: By Null String, Key Size (Bits): 1024), and "Parameters used in Phone" (SIP Phone Port: 5060). At the bottom, there are buttons for Save, Delete, Copy, Reset, and Add New. A note indicates that an asterisk (*) denotes a required item.



12.1.8 User ID List for 3rd Party Phones

Find and List Users - Microsoft Internet Explorer provided by Cisco Systems, Inc.

Address: <https://172.25.67.126:8443/ccmadmin/userFindList.do?lookup=false&multiple=true&recCnt=0&colCnt=5>

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**

Find and List Users

Status
8 records found

Search Options
Find user where: First name | begins with | **Find** Search Within Results
(firstname begins with any)

Search Results

	User ID	First Name	Last Name	Department
<input type="checkbox"/>	grand4225	Twentyfive	Mercury	
<input type="checkbox"/>	grand4224	Twentyfour	Mercury	
<input type="checkbox"/>	grand4227	Twentyseven	Mercury	
<input type="checkbox"/>	grand4226	Twentysix	Mercury	
<input type="checkbox"/>	mercury23	Twentythree	Mercury	
<input type="checkbox"/>	mercury22	Twentytwo	Mercury	
<input type="checkbox"/>	mercury20	twenty	mercury	
<input type="checkbox"/>	mercury21	twentyone	mercury	

Rows per Page: 50

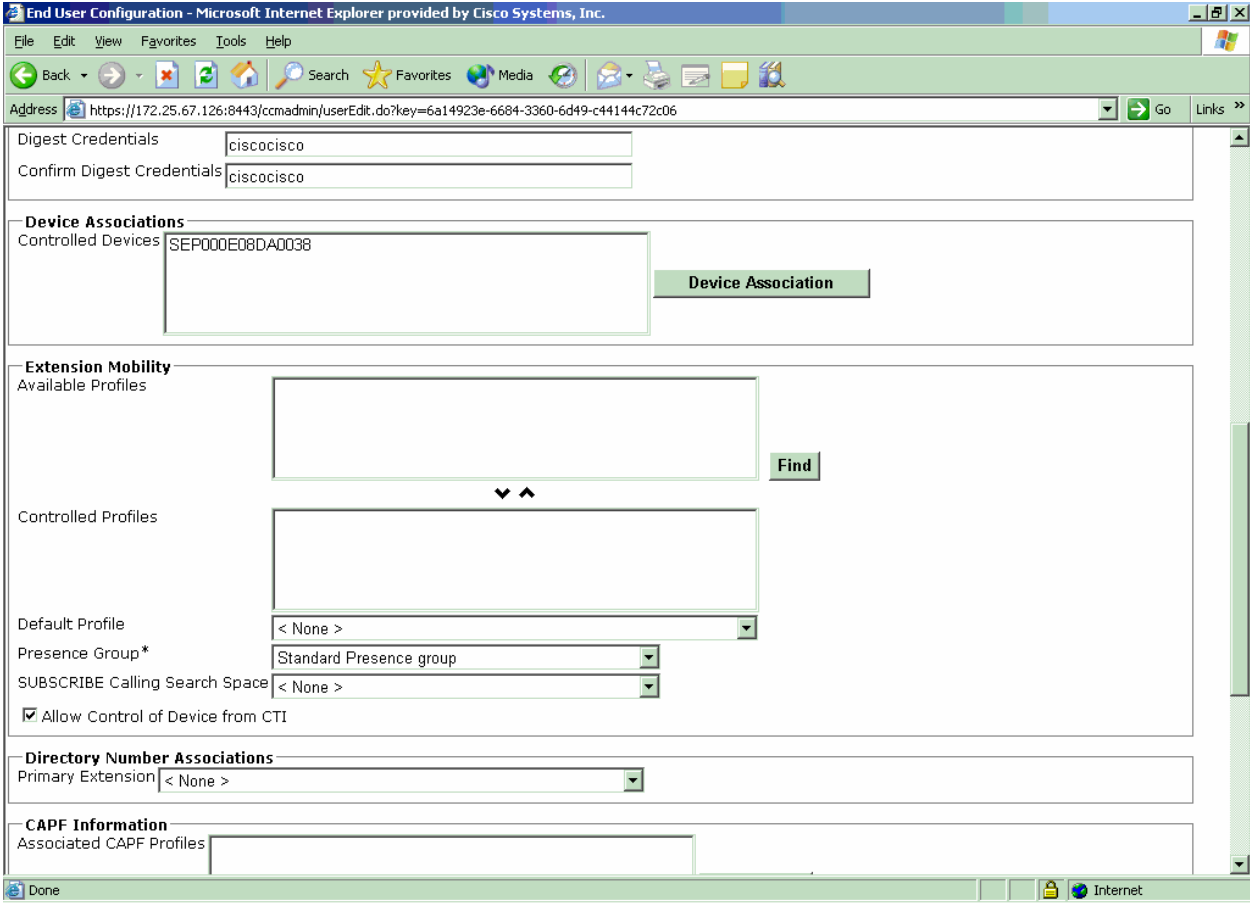


12.1.9 End User Configuration for 3rd Party Phone

The screenshot shows the Cisco CallManager Administration web interface. The browser title is "End User Configuration - Microsoft Internet Explorer provided by Cisco Systems, Inc." and the address bar shows "https://172.25.67.126:8443/ccmadmin/userSave.do". The page header includes "Cisco CallManager Administration" and "Logged in as: CCMAdministrator". A navigation menu is visible with options like System, Call Routing, Media Resources, Voice Mail, Device, Application, User Management, Bulk Administration, and Help. The main content area is titled "End User Configuration" and shows a "Status" message: "Update successful". Below this is the "User Information" section with the following fields and values:

LDAP Sync Status	Active
User ID*	mercury21
Password*
Confirm Password*
PIN*
Confirm PIN*
Last name*	mercury
Middle name	
First name	twentyone
Telephone Number	
Mail ID	
Manager User ID	
Department	
User Locale	English United States
Associated PC	





12.1.10Partitions Configuration

The screenshot shows the Cisco CallManager Administration web interface in Microsoft Internet Explorer. The browser title is "Find and List Partitions - Microsoft Internet Explorer provided by Cisco Systems, Inc." and the address bar shows the URL: https://172.25.67.126:8443/ccmadmin/partitionFindList.do?<%=reqParams%>&recCnt=0&colCnt=3. The page header includes "Cisco CallManager Administration" and "Logged in as: CCMAdministrator". A navigation menu contains items like System, Call Routing, Media Resources, Voice Mail, Device, Application, User Management, Bulk Administration, and Help. The main content area is titled "Find and List Partitions" and shows a status of "2 records found". Search options include a dropdown for "Partition Name" set to "begins with", a "Find" button, and a "Search Within Results" checkbox. The search results table lists two partitions: "Incoming Trunk" and "Phones". Below the table are buttons for "Add New", "Select All", "Clear All", and "Delete Selected", along with a "Rows per Page" dropdown set to 50.

Status
2 records found

Search Options
Find partition where Partition Name begins with Find Search Within Results
(name begins with any)

Search Results

Partition Name	Description
<input type="checkbox"/> Incoming Trunk	Incoming Trunk
<input type="checkbox"/> Phones	Phones

Add New Select All Clear All Delete Selected Rows per Page 50



12.1.11 Incoming Trunk CSS Configuration

The screenshot shows the Cisco CallManager Administration web interface in Microsoft Internet Explorer. The browser address bar shows the URL: <https://172.25.67.126:8443/ccmadmin/cssEdit.do?key=6319b4ca-1372-efb5-c287-f8c704c54b1a>. The page title is "Calling Search Space Configuration". The navigation menu includes "Cisco CallManager Administration" and "Go". The user is logged in as "CCMAdministrator". The main menu includes "System", "Call Routing", "Media Resources", "Voice Mail", "Device", "Application", "User Management", "Bulk Administration", and "Help". The "Log Off" button is visible. The "Calling Search Space Configuration" section has a "Related Links" dropdown set to "Back To Find/List" and a "Go" button. Below this is a "Status" section showing "Status: Ready". The "Calling Search Space Information" section has a "Name*" field containing "Incoming Trunk" and an empty "Description" field. The "Route Partitions for this Calling Search Space" section shows "Available Partitions" with "Phones" and "Selected Partitions (Ordered by highest priority)" with "Incoming Trunk". At the bottom, there are "Save", "Delete", "Copy", and "Add New" buttons. A note indicates that "*" indicates a required item. The browser status bar shows "Done" and "Internet".



12.1.12 Phones CSS Configuration

The screenshot shows the Cisco CallManager Administration web interface in Microsoft Internet Explorer. The browser address bar shows the URL: `https://172.25.67.126:8443/ccmadmin/cssEdit.do?key=7e73c44b-3e3b-beac-a957-221ec6a1ad2e`. The page title is "Calling Search Space Configuration - Microsoft Internet Explorer provided by Cisco Systems, Inc.". The navigation bar includes "Cisco CallManager Administration" and "Logged in as: CCMAdministrator". A menu bar contains: System, Call Routing, Media Resources, Voice Mail, Device, Application, User Management, Bulk Administration, and Help. A "Log Off" button is also present. The main content area is titled "Calling Search Space Configuration" and includes a "Related Links" section with a "Back To Find/List" button. Below this is a "Status" section showing "Status: Ready". The "Calling Search Space Information" section contains a "Name*" field with the value "Phones" and an empty "Description" field. The "Route Partitions for this Calling Search Space" section shows "Available Partitions" with "Incoming Trunk" and "Selected Partitions (Ordered by highest priority)" with "Phones". At the bottom, there are "Save", "Delete", "Copy", and "Add New" buttons, and a note: "*- indicates required item." The browser status bar at the bottom shows "Done" and "Internet".



12.1.13 Media Resource Group

The screenshot shows the Cisco CallManager Administration web interface. The browser title is "Media Resource Group Configuration - Microsoft Internet Explorer provided by Cisco Systems, Inc.". The address bar shows the URL: <https://172.25.67.126:8443/ccmadmin/mrsrcGroupEdit.do?key=1efcdd23-5b3f-e41d-7777-e33b30959a5f>. The navigation menu includes "Cisco CallManager Administration" and "Go". The user is logged in as "CCMAdministrator". The main menu includes "System", "Call Routing", "Media Resources", "Voice Mail", "Device", "Application", "User Management", "Bulk Administration", and "Help". The "Log Off" button is visible. The page title is "Media Resource Group Configuration" with a "Related Links: Back To Find/List" dropdown. The status section shows "Status: Ready". The "Media Resource Group" is "MRG-CM-MERCURY (used by 4 devices)". The "Media Resource Group Information" section has "Name * MRG-CM-MERCURY" and "Description MRG-CM-MERCURY". The "Devices for this Group" section shows "Available Media Resources **" (empty) and "Selected Media Resources *" containing "ANN_2 (ANN)", "CFB_2 (CFB)", "MOH_2 (MOH)", and "MTP_2 (MTP)". There is a checkbox for "Use Multicast for MOH Audio (If at least one multicast MOH resource is available)". At the bottom, there are buttons for "Save", "Delete", "Copy", "Reset", and "Add New".



12.1.14 Media Resource Group List

The screenshot shows the Cisco CallManager Administration web interface. The browser title is "Media Resource Group List Configuration - Microsoft Internet Explorer provided by Cisco Systems, Inc.". The address bar shows the URL: <https://172.25.67.126:8443/ccmadmin/mrsrclEdit.do?key=c63bdabb-c09a-5d53-b035-42912ca48e75>. The page header includes "Cisco CallManager Administration" and "Logged in as: CCMAdministrator". A navigation menu contains: System, Call Routing, Media Resources, Voice Mail, Device, Application, User Management, Bulk Administration, and Help. The main content area is titled "Media Resource Group List Configuration" and shows "MRGL-CM-MERCURY (used by 18 devices)". Under "Media Resource Group List Information", the "Name*" field contains "MRGL-CM-MERCURY". The "Media Resource Groups for this List" section has two lists: "Available Media Resource Groups" (empty) and "Selected Media Resource Groups" (containing "MRGL-CM-MERCURY"). At the bottom, there are buttons for "Save", "Delete", "Copy", "Reset", and "Add New". A note states: "i *- indicates required item." The status bar at the bottom shows "Done" and "Internet".



12.1.15 Voicemail Profile Configuration

The screenshot shows the Cisco CallManager Administration web interface in Microsoft Internet Explorer. The browser title is "Voice Mail Profile Configuration - Microsoft Internet Explorer provided by Cisco Systems, Inc." and the address bar shows the URL: <https://172.25.67.125:8443/ccmadmin/vmProfileEdit.do?key=65caadd1-b95a-ee9a-3e8f-8ece8ed91d61>. The page title is "Cisco CallManager Administration" with the subtitle "For Cisco IP Telecommunication Solutions". The user is logged in as "CCMAdministrator".

The main navigation menu includes: System, Call Routing, Media Resources, Voice Mail, Device, Application, User Management, Bulk Administration, and Help. A "Log Off" button is also present. The current page is "Voice Mail Profile Configuration" with a "Related Links" section containing "Back To Find/List" and a "Go" button.

The configuration form includes the following fields:

- Status:** Status: Ready
- Voice Mail Profile:** Unity2 (used by 5 devices)
- Voice Mail Profile Name *:** Unity2
- Description:** Unity Integration
- Voice Mail Pilot **:** 2904/Phones
- Voice Mail Box Mask:** (empty)
- Make this the default Voice Mail Profile for the System

Buttons at the bottom of the form are: Save, Delete, Copy, Reset, and Add New.

Help text below the form:

- * - indicates required item.
- ** The Voice Mail Pilot is comprised of the Voice Mail Pilot Number and it's corresponding Calling Search Space Name (< Voice Mail Pilot Number >/< Calling Search Space >).



12.1.16 Voicemail Pilot

The screenshot shows the Cisco CallManager Administration web interface in Microsoft Internet Explorer. The browser title is "Voice Mail Pilot Configuration - Microsoft Internet Explorer provided by Cisco Systems, Inc." and the address bar shows the URL: <https://172.25.67.125:8443/ccmadmin/vmPilotEdit.do?key=4a0dc62-cfb1-05b6-4a74-a75ed21a8d86>. The page header includes "Cisco CallManager Administration" and "Logged in as: CCMAdministrator". A navigation menu contains: System, Call Routing, Media Resources, Voice Mail, Device, Application, User Management, Bulk Administration, and Help. A "Log Off" button is also present. The main content area is titled "Voice Mail Pilot Configuration" and includes a "Related Links" section with a "Back To Find/List" button. Below this is a "Status" section showing "Status: Ready". The "Voice Mail Pilot Information" section contains the following fields: "Voice Mail Pilot Number" (text input with value 2904), "Calling Search Space" (dropdown menu with value Phones), and "Description" (text input with value Unity 2 Integration). There is a checkbox labeled "Make this the default Voice Mail Pilot for the system" which is currently unchecked. At the bottom of the form are buttons for "Save", "Delete", and "Add New". A note below the buttons states: "i *- indicates required item." The browser status bar at the bottom shows "Done" and "Internet".

12.1.17 Voicemail Ports

The screenshot shows the Cisco CallManager Administration web interface. The browser title is "Find and List Voice Mail Ports - Microsoft Internet Explorer provided by Cisco Systems, Inc.". The address bar shows the URL: `https://172.25.67.125:8443/ccmadmin/vmPortFindList.do?lookup=false&multiple=true&recCnt=0&colCnt=7`. The page header includes "Cisco CallManager Administration" and "Logged in as: CCMAdministrator". A navigation menu is visible with options like System, Call Routing, Media Resources, Voice Mail, Device, Application, User Management, Bulk Administration, and Help. The main content area is titled "Find and List Voice Mail Ports" and shows a status of "2 records found". Below this is a search options section with a dropdown for "Device Name" set to "begins with" and a "Find" button. The search results are displayed in a table with columns: Device Name, Description, Device Pool, SCCP Security Profile, Status, and IP Address. Two records are shown: "Unity2-VI1" and "Unity2-VI2", both with status "Registered with CM-VENUS" and IP address "172.20.214.250". At the bottom of the results table are buttons for "Add New", "Select All", "Clear All", "Delete Selected", and "Reset Selected", along with a "Rows per Page" dropdown set to 50.

Status
2 records found

Search Options
Find Voice Mail Port where begins with Search Within Results

(device.name begins with any)

Search Results

Device Name	Description	Device Pool	SCCP Security Profile	Status	IP Address	Copy
<input type="checkbox"/> Unity2-VI1	Unity Integration	Default	Standard SCCP Profile for Auto Registration	Registered with CM-VENUS	172.20.214.250	
<input type="checkbox"/> Unity2-VI2	Unity Integration	Default	Standard SCCP Profile for Auto Registration	Registered with CM-VENUS	172.20.214.250	

Rows per Page



Voice Mail Port Configuration - Microsoft Internet Explorer provided by Cisco Systems, Inc.

File Edit View Favorites Tools Help

Address <https://172.25.67.125:8443/ccmadmin/vmPortEdit.do?key=feeee958-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-V11
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

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

The screenshot shows a web browser window displaying the Cisco CallManager Administration interface. The browser's address bar shows the URL: `https://172.25.67.125:8443/ccmadmin/messageWaitingEdit.do?key=a29e6333-68d9-4d2e-1003-c2d8cbb4fada`. The page title is "Message Waiting Configuration - Microsoft Internet Explorer provided by Cisco Systems, Inc.". The interface includes a navigation menu with options like "System", "Call Routing", "Media Resources", "Voice Mail", "Device", "Application", "User Management", "Bulk Administration", and "Help". The user is logged in as "CCMAdministrator". The main content area is titled "Message Waiting Configuration" and shows the following configuration details:

- Status:** Status: Ready
- Message Waiting Information:**
 - Message Waiting Number: *2999
 - Partition: Phones
 - Description: Unity Integration
 - Message Waiting Indicator: On Off
 - Calling Search Space: Phones

At the bottom of the configuration area, there are buttons for "Save", "Delete", "Copy", and "Add New". A note below the buttons states: "i *- indicates required item."



The screenshot shows a web browser window titled "Message Waiting Configuration - Microsoft Internet Explorer provided by Cisco Systems, Inc.". The address bar contains the URL: <https://172.25.67.125:8443/ccmadmin/messageWaitingEdit.do?key=24a6a354-3cd1-4f18-1acf-a5fed4d54161>. The page header includes "Cisco CallManager Administration" and "Logged in as: CCMAdministrator". A navigation menu lists various system components like "System", "Call Routing", "Media Resources", "Voice Mail", "Device", "Application", "User Management", "Bulk Administration", and "Help". The main content area is titled "Message Waiting Configuration" and includes a "Status" section showing "Status: Ready". Below this is the "Message Waiting Information" section with the following fields: "Message Waiting Number" (required, value: 999), "Partition" (value: Phones), "Description" (value: Unity Integration), "Message Waiting Indicator" (radio buttons for "On" and "Off", with "Off" selected), and "Calling Search Space" (value: Phones). At the bottom of the form are buttons for "Save", "Delete", "Copy", and "Add New". A note indicates that an asterisk (*) denotes a required item. The browser's status bar at the bottom shows "Done" and "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)

Provide Outside Dial Tone 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

The screenshot shows a web browser window with the following content:

- Association Information:**
 - Modify Button Items
 - 1 Line [1] - 4224 in Phones
 - Add On Module(s) -----
 - 2 Line [2] - Add a new DN
 - Unassigned Associated Items -----
 - 3 Add a new SD
 - 4 Privacy
 - 5 None
- Phone Type:**
 - Product Type: Third-party SIP Device (Basic)
 - Device Protocol: SIP
- Device Information:**
 - Registration: Registered with Cisco CallManager CM-MERCURY
 - IP Address: 172.20.236.19
 - MAC Address*: 000B8202679F
 - Description: Grandstream 100 SIP 4224
 - Device Pool*: Default
 - Phone Button Template*: Third-party SIP Device (Basic)
 - Common Phone Profile*: Standard Common Phone Profile
 - Calling Search Space: < None >
 - 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

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

Address: <https://172.25.67.126:8443/ccmadmin/phoneEdit.do?key=5ff88d08-cc9d-9340-4e5a-13777256d8cf>

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*: 711 ulaw
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.



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

Address: <https://172.25.67.126:8443/ccmadmin/directoryNumberEdit.do?key=6cfd3487-a069-7149-9399-8880db926be88mapkey=9a937a5b-e884-3c3c-b376-0e895904a003&devi>

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

Forward All or Voice Mail Destination

Calling Search Space Phones

Secondary Calling Search Space for Forward All < None >

Find



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

Address: https://172.25.67.126:8443/ccmadmin/directoryNumberEdit.do?key=6cfd3487-a069-7149-9399-8880db926be88mapkey=9a937a5b-e884-3c3c-b376-0e895904a003&devi

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)



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

Address: <https://172.25.67.126:8443/ccmadmin/directoryNumberEdit.do?key=6cfd3487-a069-7149-9399-8880db926be88mapkey=9a937a5b-e884-3c3c-b376-0e895904a003&devi>

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): 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):

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*:

Busy Trigger*: (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.



12.2.2 Grandstream GXP-2000

The screenshot shows a Microsoft Internet Explorer browser window displaying the Cisco Phone Configuration page. The address bar shows the URL: <https://172.25.67.126:8443/ccmadmin/phoneEdit.do?key=e6ce436e-6b11-1a88-e2e2-decf33f3063>. The page content is organized into several sections:

- Association Information:** A list of 11 lines. Line 1 is selected and labeled "Line [1] - 4226 in Phones". Other lines are labeled "Line [2] - 4236 in Phones" through "Line [8] - Add a new DN". Lines 10 and 11 are labeled "Privacy" and "None" respectively. A "Modify Button Items" button is visible at the top of this section.
- Phone Type:** Product Type: **Third-party SIP Device (Advanced)**; Device Protocol: **SIP**.
- Device Information:** Registration: Partial Registered; IP Address: 172.20.215.126; MAC Address*: 000B82073374; Description: Grandstream GXP2000 SIP 4226; Device Pool*: Default; Phone Button Template*: Third-party SIP Device (Advanced); Common Phone Profile*: Standard Common Phone Profile; Calling Search Space: < None >; Media Resource Group List: MRGL-CM-MERCURY; Location*: Hub_None; Owner User ID: < None >. Checkboxes include "Retry Video Call as Audio" (checked) and "Ignore Presentation Indicators (internal calls only)" (unchecked).
- 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. A checkbox for "Media Termination Point Required" is checked.



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

File Edit View Favorites Tools Help

Address <https://172.25.67.126:8443/ccmadmin/phoneEdit.do?key=e6ce436e-6b11-1a88-e2e2-dec9f33f3063> 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

i *- indicates required item.
i - 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.

Address: https://172.25.67.126:8443/ccmadmin/directoryNumberEdit.do?key=ac2c6bc9-56f7-8157-008c-77aa6f33425e&mapkey=1a08b3d3-c74e-107f-fc15-6211d2577597&device

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

Forward All or Voice Mail Destination Calling Search Space

or < None >



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

File Edit View Favorites Tools Help

Address: https://172.25.67.126:8443/ccmadmin/directoryNumberEdit.do?key=ac2c6bc9-56f7-8157-008c-77aa6f33425e&mapkey=1a08b3d3-c74e-107f-fc15-6211d2577597&device

Voice Mail Destination Calling Search Space

Forward All or < None >

Secondary Calling Search Space for Forward All < None > **Find**

Forward Busy Internal or < None >

Forward Busy External or < None >

Forward No Answer Internal or < None >

Forward No Answer External or < None >

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 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.

Address: <https://172.25.67.126:8443/ccmadmin/directoryNumberEdit.do?key=ac2c6bc9-56f7-8157-008c-77aa6f33425e&mapkey=1a08b3d3-c74e-107f-fc15-6211d2577597&device>

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): 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):

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*:

Busy Trigger*: (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.



12.3 GrandStream Budgetone 100 SIP phone webpage Configuration

The screenshot shows a web browser window titled "Grandstream Device Configuration - Microsoft Internet Explorer". The address bar shows "http://172.20.236.19/index.htm". The page content is titled "Grandstream Device Configuration" and has three tabs: "STATUS", "BASIC SETTINGS", and "ADVANCED SETTINGS". The "STATUS" tab is selected, displaying the following information:

- MAC Address:** 00.0B.82.02.67.9F
- IP Address:** 172.20.236.19
- Product Model:** BT100 REV 2.0
- Software Version:** Program-- 1.0.8.16 Bootloader-- 1.0.8.9 HTML-- 1.0.8.16
VOC-- 1.0.1.0
- System Up Time:** 1 day(s) 4 hour(s) 20 minute(s)
- Registered:** Yes
- PPPoE Link Up:** disabled
- NAT:**
 - NAT Mapped IP:** 0.0.0.0
 - NAT Mapped Port:** 0
- Total Inbound Calls:** 6
- Total Outbound Calls:** 19
- Total Missed Calls:** 3
- Total Call Time (in minutes):** 63
- Total SIP Message Sent:** 1204
- Total SIP Message Received:** 2260
- Total RTP Packet Sent:** 126732
- Total RTP Packet Received:** 126941
- Total RTP Packet Loss:** 0

The browser status bar at the bottom shows "Done" and "Local intranet".

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:

statically configured as:

IP Address:

Subnet Mask:

Default Router:

DNS Server 1:

DNS Server 2:

Time Zone:

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

All Rights Reserved Grandstream Networks, Inc. 2005



Grandstream Device Configuration - Microsoft Internet Explorer

File Edit View Favorites Tools Help

Back Forward Stop Refresh Home Search Favorites Media Print

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: (e.g., sip.mycompany.com, or IP address)

Outbound Proxy: (e.g., proxy.myprovider.com, or IP address, if any)

SIP User ID: (the user part of an SIP address)

Authenticate ID: (can be identical to or different from SIP User ID)

Authenticate Password: (purposely not displayed for security protection)

Name: (optional, e.g., John Doe)

Advanced Options:

Preferred Vocoder: (in listed order)

choice 1:

choice 2:

choice 3:

choice 4:

choice 5:

choice 6:

choice 7:

choice 8:

G723 rate: 6.3kbps encoding rate 5.3kbps encoding rate

iLBC frame size: 20ms 30ms

iLBC payload type: (between 96 and 127, default is 97)

Done Local intranet

Grandstream Device Configuration - Microsoft Internet Explorer

File Edit View Favorites Tools Help

Back Forward Stop Refresh Home Search Favorites Media Print

Address http://172.20.236.19/config.htm Go Links

LLBC payload type: (between 96 and 127, default is 97)

Silence Suppression: No Yes

Voice Frames per TX: (up to 10/20/32/64 for G711/G726/G723/other codecs respectively)

Layer 3 QoS: (Diff-Serv or Precedence value)

Layer 2 QoS: 802.1Q/VLAN Tag 802.1p priority value (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: (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: (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: (default 5060)

local RTP port: (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 Forward Stop Refresh Home Search Favorites Media Print

Address <http://172.20.236.19/config.htm> Go Links

keep-alive interval: (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:

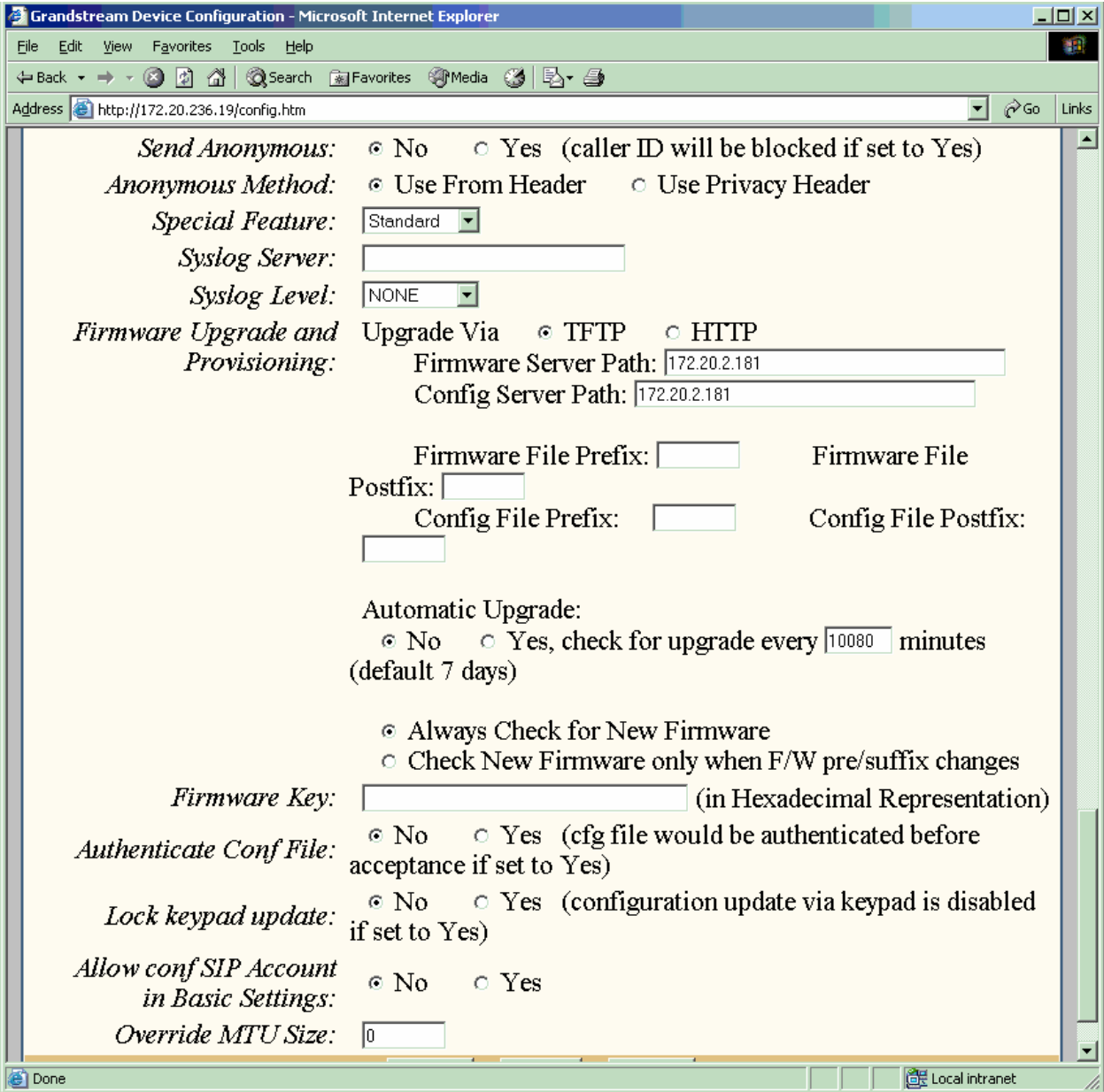
Send Flash Event: No Yes (Flash will be sent as a DTMF event if set to Yes)

Onhook Threshold:

NTP Server: (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



12.4 GrandStream GXP-2000 SIP phone webpage Configuration

Grandstream Device Configuration - Microsoft Internet Explorer

Address <http://172.20.215.126/index.htm>

Grandstream Device Configuration

STATUS	BASIC SETTINGS	ADVANCED SETTINGS	ACCOUNT 1	ACCOUNT 2	ACCOUNT 3	ACCOUNT 4
MAC Address: 00.0B.82.07.33.74						
IP Address: 172.20.215.126						
Product Model: GXP2000						
Software Version: Program-- 1.0.2.3 Bootloader-- 1.0.2.3						
System Up Time: 0 day(s) 5 hour(s) 49 minute(s)						
Registered: Account 1: Yes Account 2: Yes Account 3: No Account 4: No						
PPPoE Link Up: disabled detected NAT type is open Internet						

All Rights Reserved Grandstream Networks, Inc. 2004, 2005

Local intranet

Grandstream Device Configuration

STATUS BASIC SETTINGS ADVANCED SETTINGS ACCOUNT 1 ACCOUNT 2 ACCOUNT 3 ACCOUNT 4

End User Password: (purposefully 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:

statically configured as:

IP Address:

Subnet Mask:

Default Router:

DNS Server 1:

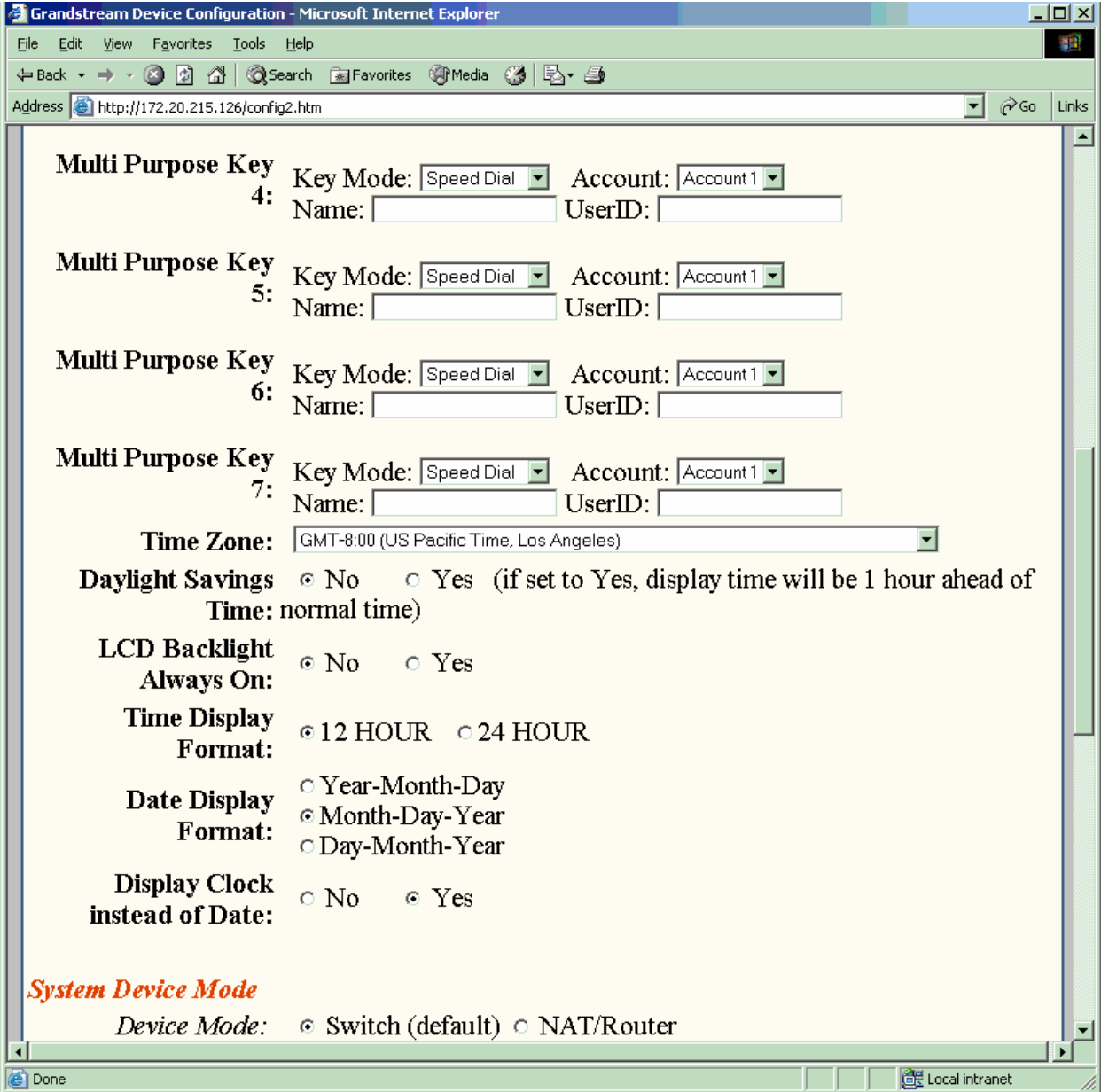
DNS Server 2:

Multi Purpose Key 1: Key Mode: Account:
Name: UserID:

Multi Purpose Key 2: Key Mode: Account:
Name: UserID:

Multi Purpose Key 3: Key Mode: Account:
Name: UserID:





System Device Mode
Device Mode: Switch (default) NAT/Router

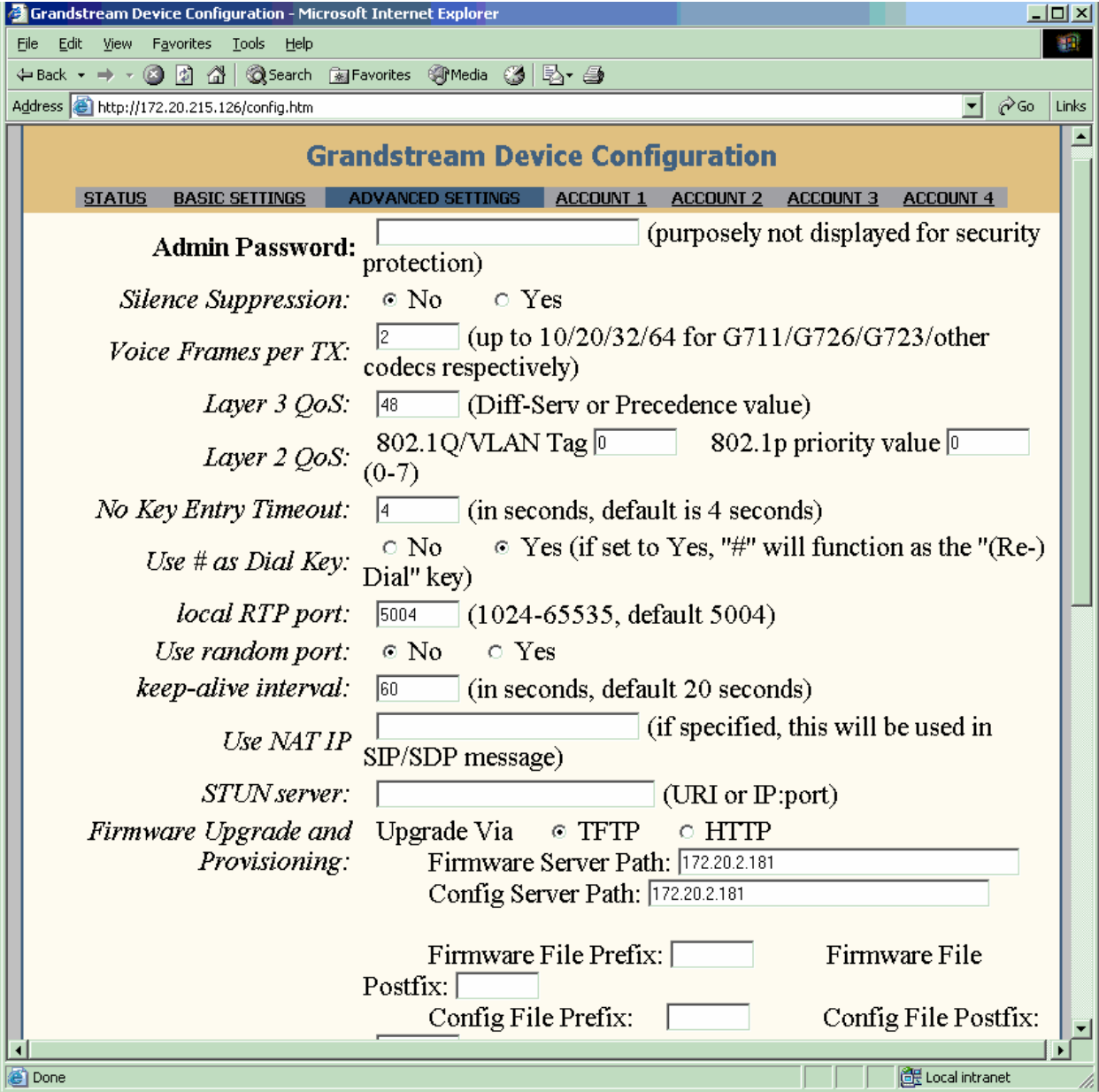
NAT/Router Configuration
WAN side http access: No Yes (WAN side access to http server will be rejected if set to No)
Reply to ICMP on WAN port: No Yes (Unit will not respond to PING from WAN side if set to No)
Cloned WAN MAC Addr: (in hex format)
LAN Subnet Mask: (default is 255.255.255.0)
LAN DHCP Base 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:

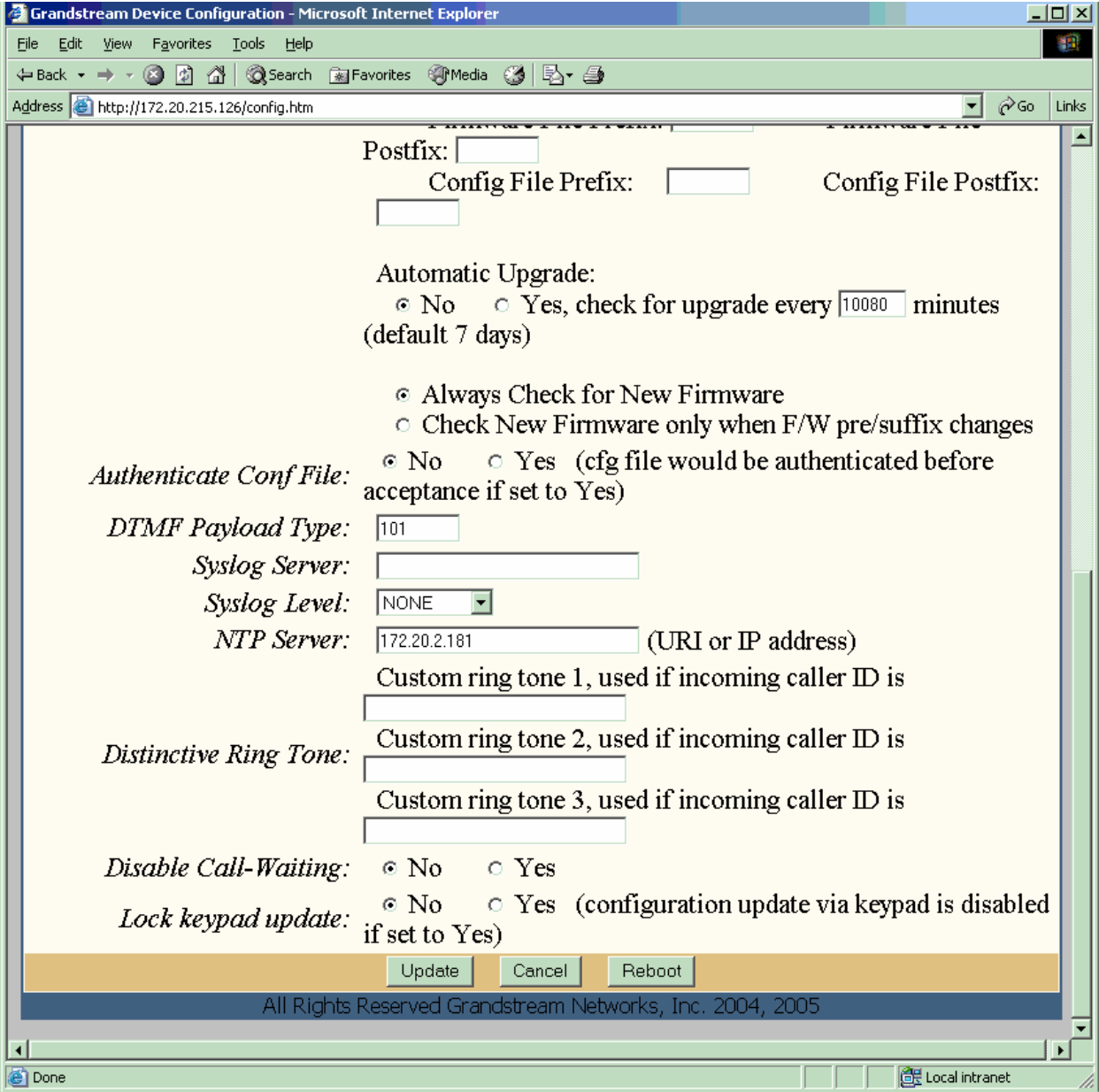
WAN port	LAN IP	LAN port	Protocol
<input type="text"/> 0	<input type="text"/>	<input type="text"/> 0	UDP Only
<input type="text"/> 0	<input type="text"/>	<input type="text"/> 0	UDP Only
<input type="text"/> 0	<input type="text"/>	<input type="text"/> 0	UDP Only
<input type="text"/> 0	<input type="text"/>	<input type="text"/> 0	UDP Only
<input type="text"/> 0	<input type="text"/>	<input type="text"/> 0	UDP Only
<input type="text"/> 0	<input type="text"/>	<input type="text"/> 0	UDP Only
<input type="text"/> 0	<input type="text"/>	<input type="text"/> 0	UDP Only
<input type="text"/> 0	<input type="text"/>	<input type="text"/> 0	UDP Only
<input type="text"/> 0	<input type="text"/>	<input type="text"/> 0	UDP Only

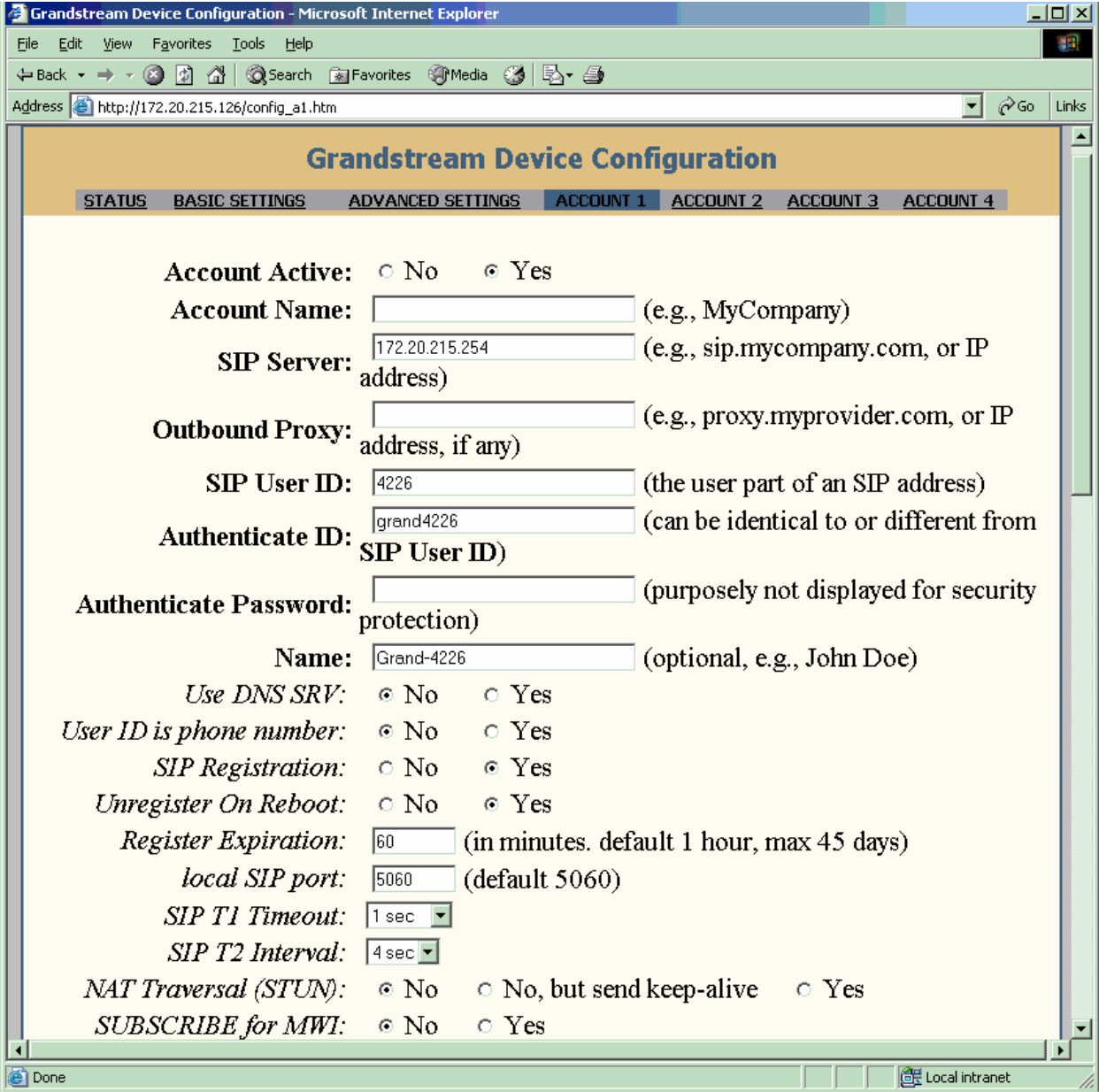
Port Forwarding:

Update Cancel Reboot









Grandstream Device Configuration - Microsoft Internet Explorer

File Edit View Favorites Tools Help

Back Forward Stop Refresh Home Search Favorites Media Print

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: (in seconds. default 180 seconds)

Min-SE: (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)

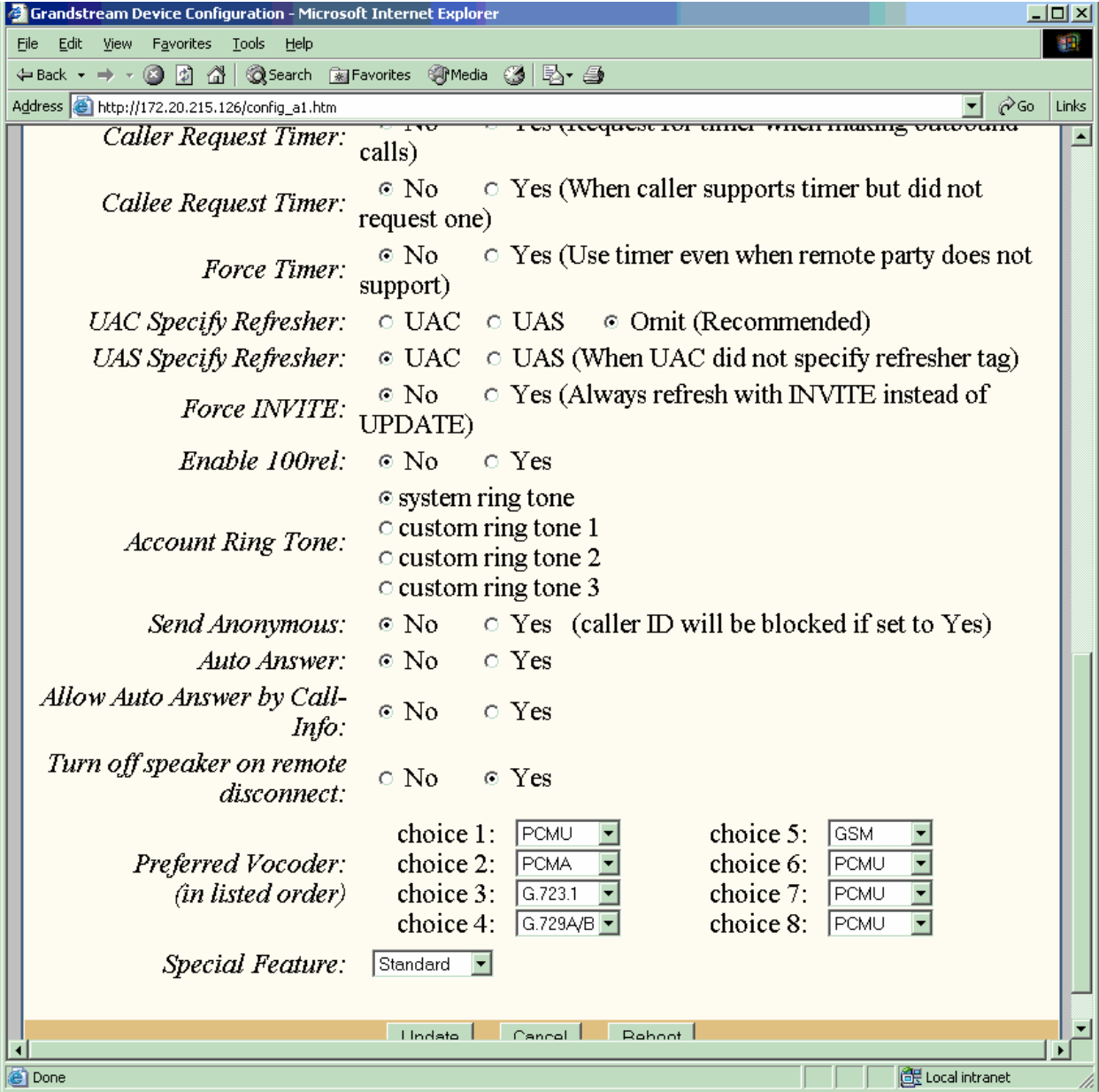
Force INVITE: No Yes (Always refresh with INVITE instead of UPDATE)

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 Forward Stop Refresh Home Search Favorites Media Print

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: (e.g., sip.mycompany.com, or IP address)

Outbound Proxy: (e.g., proxy.myprovider.com, or IP address, if any)

SIP User ID: (the user part of an SIP address)

Authenticate ID: (can be identical to or different from SIP User ID)

Authenticate Password: (purposely not displayed for security protection)

Name: (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: (in minutes. default 1 hour, max 45 days)

local SIP port: (default 5062)

SIP T1 Timeout:

SIP T2 Interval:

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 Forward Stop Refresh Home Search Favorites Media Print

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: (in seconds. default 180 seconds)

Min-SE: (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

