

## Application Notes

### Grandstream GXP2000 / GXP2020 and VCX v7.1.11c and 7.2.56.56

**Issue:** Revision 1.4

**Date:** June 20, 2007

**Abstract:** This application note describes the configuration procedures required to configure the Grandstream GXP2000 4-line Enterprise SIP phone and GXP2020 6-line Enterprise SIP Phone with the 3Com VCX platform. The GXP2000 and GXP2020 share the same SIP stack and F/W.

The GXP Series offers power-over-Ethernet (PoE), dual 10/100 auto-sensing Ethernet ports, multi-lines with individual SIP accounts and XML capabilities. The GXP Series is expandable using a 56 button expansion car, offers MLS (multi-language support) in 5 languages, AES encryption and remote configuration for quick deployment over a large network.

The GXP2020 offers more advanced features than the GXP2000 including 4 programmable XML capable buttons, 6 lines each with individual SIP accounts, 5-way conferencing, two headset jacks (RJ22/2.5mm) and a large (320 x 160 pixels) high-resolution backlit LCD display and an elegant new design.

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**Revision History**

Revision	Date	Author	Reason for change
1.0	25 Apr 07	Marianne Rocco & Soumil Vora	Initial revision
1.1	4 June 07	Marianne Rocco & Soumil Vora	Added tested observations
1.3	6 June 07	Marianne Rocco & Soumil Vora	Added topology map, marketing materials and new test cases
1.4	6 June 07	Bob Blair	Formatting, Added 3Com Specific data

**References**

Date	Document Name	Revision	Company
25 Apr 07	GXP User's Manual GXP Release Notes		Grandstream Networks
25 Apr 07	GXP2020 Release Notes		Grandstream Networks

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## Objective

Configure the Grandstream Enterprise SIP Phones (GXP2000, GXP2020 and GXV3000) for use with the 3COM VCX IP Call Processor v7.1.11c and 7.2.56.56.

### ***Grandstream Networks Company and Product Details***

Grandstream Networks is a leading designer and manufacturer of innovative, affordable, and high quality IP voice and video products for the worldwide broadband telephony market. Our products are fully compatible with the SIP industry standard, field proven with large and rapidly growing deployed base, and have broad interoperability with the majority of 3rd party SIP products on the market today.

The Grandstream IP voice & video products offer the best price-performance point in the industry. Each is based on SIP standard and is feature rich – supporting both traditional and advanced features - support a broad range of voice codecs, and are easy to manage and deploy through web-based GUI interfaces.

Grandstream continues to bring innovation to the IP communications market with exciting products of compelling values and differentiations. Grandstream Networks is headquartered in Brookline, Massachusetts with offices in Dallas, Los Angeles and Shenzhen/China.

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**GXP2000 Features at a Glance**

<b>Open Standards Compatible</b>	SIP 2.0, TCP/IP/UDP, RTP/RTCP, HTTP/HTTPS, ARP/RARP, ICMP, DNS (A record and SRV), DHCP (both client and server), PPPoE, PoE, TFTP, NTP, Telnet, and TLS.
<b>Network Interfaces</b>	Dual 10/100Mbps Ethernet ports; 2 USB (2.0) host ports, headset jack (2.5mm).
<b>Superb Audio Quality</b>	Advanced Digital Signal Processing (DSP), Silence suppression, VAD, CNG, AGC.
<b>Feature Rich</b>	Traditional voice features including caller ID, call waiting, hold, transfer, forward, block, mute, autodial, off-hook dial, and click to dial.
<b>Advanced Features</b>	Tilt screen, multi-line support, 5 navigation keys, duplex speakerphone/hands-free, individual SIP accounts, multi-language support (MLS), XML enabled, SRTP & TLS
<b>Advanced Functionality</b>	Custom down-loadable ring-tones, multi-line support, multi-party conferencing, up to 112 additional extensions with side-car, headset enabled, intercom, AES encryption.

**GXP2020 Features at a Glance**

<b>Open Standards Compatible</b>	SIP 2.0, TCP/IP/UDP, RTP/RTCP, HTTP/HTTPS, ARP/RARP, ICMP, DNS (A record and SRV), DHCP (both client and server), PPPoE, PoE, TFTP, NTP, Telnet, and TLS.
<b>Superb Audio Quality</b>	Advanced Digital Signal Processing (DSP), Silence suppression, VAD, CNG, AGC.
<b>Network Interfaces</b>	Dual 10/100Mbps Ethernet ports; 2 USB (2.0) host ports, 2 headset jacks (RJ22 and 2.5mm jack).
<b>Feature Rich</b>	Traditional voice features including caller ID, call waiting, hold, transfer, forward, block, autodial, off-hook dial, and click to dial.
<b>Advanced Features</b>	Multi-line support with dual-color LED, multi-party conferencing, line extension interface, large back-lit graphic LCD, 5 navigation keys, eleven (11) dedicated buttons for hold, send, speakerphone, headset, transfer, conference (for up to 4 party), mute, message, do-not-disturb, phone book, intercom/paging.
<b>Advanced Functionality</b>	Custom down-loadable ring-tones, TLS & SRTP, multi-language support, XML soft keys, 3 adjustable positioning angles, wall mountable, AES encryption.

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## Technical Summary for GXP2000 and GXP2020

### GXP2000 Technical Specifications

<b>Lines</b>	4 direct lines with 7 speed dial keys; up to 11 line calls (with an additional 112 lines with 2 daisy-chained GXP-2000 Ext)
<b>Protocol Support</b>	Support SIP 2.0, TCP/UDP/IP, PPPoE, RTP/RTCP, SRTP by SDES, HTTP, ARP/RARP, ICMP, DNS, DHCP, NTP/SNTP, TFTP, SIMPLE/PRESENCE protocols. Support multiple SIP accounts and up to 11 media channels concurrently Support SIP PUBLISH method (RFC 3903), SIP Presence package (RFC 3856, 3863) for use of 7 MFks and GXP-2000EXT, SIP Dialog package (RFC 4235) Support for SIP MESSAGE method (RFC 3428)S Stores up to 100 incoming IM messages (drops IM message 101 plus)
<b>Display</b>	8 line x 22character, 64 rows x 130 column in pixels
<b>Feature Keys</b>	8 dedicated keys: Message Button, Hold, Transfer, Conference, Speakerphone, Send, Mute/Del, 5 display/menu navigation keys, dual color LEDs
<b>Device Management</b>	NAT-friendly remote software upgrade (via TFTP/HTTP) for deployed devices including behind firewall/NAT, Auto/manual provisioning system, GUI Interface, Address Book Support Layer 2 (802.1Q, VLAN, 802.1p) and Layer 3 QoS (ToS, DiffServ, MPLS) Support for GXP-2000 Extension console and diagnostic mode for keys on GXP-2000 Extension console
<b>Audio Features</b>	Full-duplex hands-free speakerphone, headset enabled Advanced Digital Signal Processing (DSP) Dynamic negotiation of codec and voice payload length Support for G.723,1 (6.3K), G.729A/B, G.711 $\mu$ /A, G.726 (40K/32K/24K/16K), G.728, G.722 (wide-band), GSM and iLBC codecs In-band and out-of-band DTMF (in audio, RFC2833, SIP INFO) Silence Suppression, VAD (voice activity detection), CNG (comfort noise generation), ANG (automatic gain control) Acoustic Echo Cancellation (AEC) with Acoustic Gain Control (AGC) for speakerphone mode Support side tone, Adaptive jitter buffer control (patent-pending) and packet delay & loss concealment
<b>Telephony Features</b>	Intuitive graphic user interface (GUI), downloadable phone book (XML, LDAP), MLS (multi language support) Voice mail indicator with indicator, downloadable custom ring-tones Call hold, call transfer (attended/blind) Do-Not-Disturb (DND), call forwarding, call waiting, call waiting caller ID, mute, redial, call log, and volume control, caller ID display or block Transfer, hold, forward, multi-party conferencing, dial plans, off-hook auto dial, auto answer, early dial and speed dial. Support for anonymous call using privacy header
<b>Network and Provisioning</b>	Via keypad/LCD, Web browser, or secure (AES encrypted) central configuration file, manual or dynamic host configuration protocol (DHCP) network setup Support NAT traversal using IETF STUN and Symmetric RTP Support for IEEE 802.1p/Q tagging (VLAN), Layer 3 TOS
<b>Firmware Upgrades</b>	Support firmware upgrade via TFTP or HTTP, Support for Authenticating configuration file before accepting changes User specific URL for configuration file and firmware files
<b>Advanced Server Features</b>	Message waiting indication, support DNS SRV Look up and SIP Server Fail Over, Support customizable idle screen via downloading XML by HTTP/TFTP

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**Security**                      DIGEST authentication and encryption using MD5 and MD5-sess, SRTP over TLS

### GXP2020 Technical Specifications

<b>Lines</b>	6 direct lines with independent SIP accounts, 7 programmable speed dial keys, 4 XML programmable soft-keys
<b>Protocol Support</b>	Support SIP 2.0, TCP/UDP/IP, PPPoE, RTP/RTCP, SRTP by SDES, HTTP, ARP/RARP, ICMP, DNS, DHCP, NTP/SNTP, TFTP, SIMPLE/PRESENCE protocols Support multiple SIP accounts and up to 11 media channels concurrently Support SIP PUBLISH method (RFC 3903), SIP Presence package (RFC 3856, 3863) for use of 7 MFks SIP Dialog package (RFC 4235) Support for SIP MESSAGE method (RFC 3428) Stores up to 100 incoming IM messages (drops IM message 101 plus)
<b>Display</b>	Back-lit 240x160 graphic LCD with 32-level grey scales 8 line x 22character, 64 rows x 130 column in pixels
<b>Feature Keys</b>	11 dedicated keys: hold, send, speakerphone, headset, transfer, conference (up to 4 parties), mute, message, Do-not-disturb, phone book, intercom/paging, 5 display/menu navigation keys, dual color LEDs
<b>Device Management</b>	NAT-friendly remote software upgrade (via TFTP/HTTP) for deployed devices including behind firewall/NAT, Auto/manual provisioning system, GUI Interface, Address Book Support Layer 2 (802.1Q, VLAN, 802.1p) and Layer 3 QoS (ToS, DiffServ, MPLS) Key expansion interface
<b>Audio Features</b>	Full-duplex hands-free speakerphone, headset enabled. Advanced Digital Signal Processing (DSP). Dynamic negotiation of codec and voice payload length Support for G.723,1 (6.3K), G.729A/B, G.711 $\mu$ A, G.726 (40K/32K/24K/16K), G.722 (wide-band), GSM and iLBC codecs. In-band and out-of-band DTMF (in audio, RFC2833, SIP INFO) Silence Suppression, VAD (voice activity detection), CNG (comfort noise generation), ANG (automatic gain control) Acoustic Echo Cancellation (AEC) with Acoustic Gain Control (AGC) for speakerphone mode Adaptive jitter buffer control (patent-pending) and packet delay & loss concealment, Support side tone
<b>Telephony Features</b>	Intuitive graphic user interface (GUI), downloadable phone book (XML, LDAP), support for anonymous call using privacy header, MLS (multi language support) Voice mail indicator with indicator, downloadable custom ring-tones, call hold, call transfer (attended/blind), Do-Not-Disturb (DND), call forwarding, call waiting, caller ID, mute, redial, call log, and volume control, caller ID display or block Multi-party conferencing (up to 4), dial plans, off-hook auto dial, auto answer, early dial and speed dial
<b>Network and Provisioning</b>	Via keypad/LCD, Web browser, or secure (AES encrypted) central configuration file, manual or dynamic host configuration protocol (DHCP) network setup Support NAT traversal using IETF STUN and Symmetric RTP Support for IEEE 802.1p/Q tagging (VLAN), Layer 3 TOS
<b>Firmware Upgrades</b>	Support firmware upgrade via TFTP or HTTP, Support for Authenticating configuration file before accepting changes. User specific URL for configuration file and firmware files
<b>Advanced Server Features</b>	Message waiting indication, support DNS SRV Look up and SIP Server Fail Over, Support customizable idle screen via downloading XML by HTTP/TFTP
<b>Security</b>	DIGEST authentication and encryption using MD5 and MD5-sess, SRTP over TLS

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### Datasheets

- Datasheet  
<http://www.grandstream.com/resources.html> (GXP2000)  
<http://www.grandstream.com/gxp2020.html>
- Features, Functions, and Benefits  
<http://www.grandstream.com/gxp2020.html>  
<http://www.grandstream.com/gxp2000.html>



**GXP2000**



**GXP2000 w/Extension  
Module**



**GXP2020**

Picture 1: Grandstream GXP Product Photos

### ***Grandstream Networks Overview***

Grandstream designs and manufactures IP terminal endpoints for broadband networks. Grandstream IP products enable businesses to create a competitive advantage when selling a total IP Solution. They are simple to install, easy to manage, and offer advanced feature sets. Our price competitive products have recently been certified with leading SIP platforms and service providers. Grandstream SIP phones will help maximize your margins without compromising the quality of the 3Com VCX solution.

The Grandstream product portfolio includes Consumer ATAs and IP Phones, Enterprise Multi-line SIP phones (PoE powered), Analog FXO/FXS Gateways, and IP Video Phones. For more information, please visit [www.grandstream.com](http://www.grandstream.com). Grandstream Networks is headquartered in Brookline, Massachusetts with offices in Dallas, Los Angeles and Shenzhen/China.



## Configuration Technical Details

Grandstream Enterprise SIP Phones are standards-compliant SIP phones which are interoperable with most 3<sup>rd</sup> party SIP platforms, service providers, and traditional and IP-PBXs. The Grandstream SIP Phones interoperate easily with the 3Com VCX platform.

### *How it Works*

Grandstream Enterprise SIP Phones require a TFTP, FTP or an HTTP server to download firmware and configuration files. All configuration and provisioning information is detailed in these files. The standard web-GUI interface for each model guides the end-user through individual line configuration and basic / advanced set-up requirements.

### Hardware Revisions

- Grandstream GXP Series HV 1.1
- VCX V7000 IBM 306m Platform
- VCX Series i IBM 520 Server Platform
- 3102B Hard Phones
- 3COM PRI Digital Gateway V7122
- 3COM 5500G-EI Gigabit Ethernet Switch
- 3COM IP Telecommuting Module V7005

### Software Revisions

- Grandstream GXP Series Firmware release 1.1.4.8
- VCX v7.1.11c
- VCX 7.2.56.56 with IBM i5R4 O.S.
- Convergence Client v2.4 App3.1 Client-9
- 3COM OS v3.03.00s168c03 5500G Firmware
- 3CTM v4.4.3

## Installation Overview

Installation of Grandstream phones requires configuring the 3COM platform for additional phone lines/users, creating user-specific Grandstream configuration files, and provisioning the Grandstream phones with appropriate Grandstream configuration files.

## Network Topology

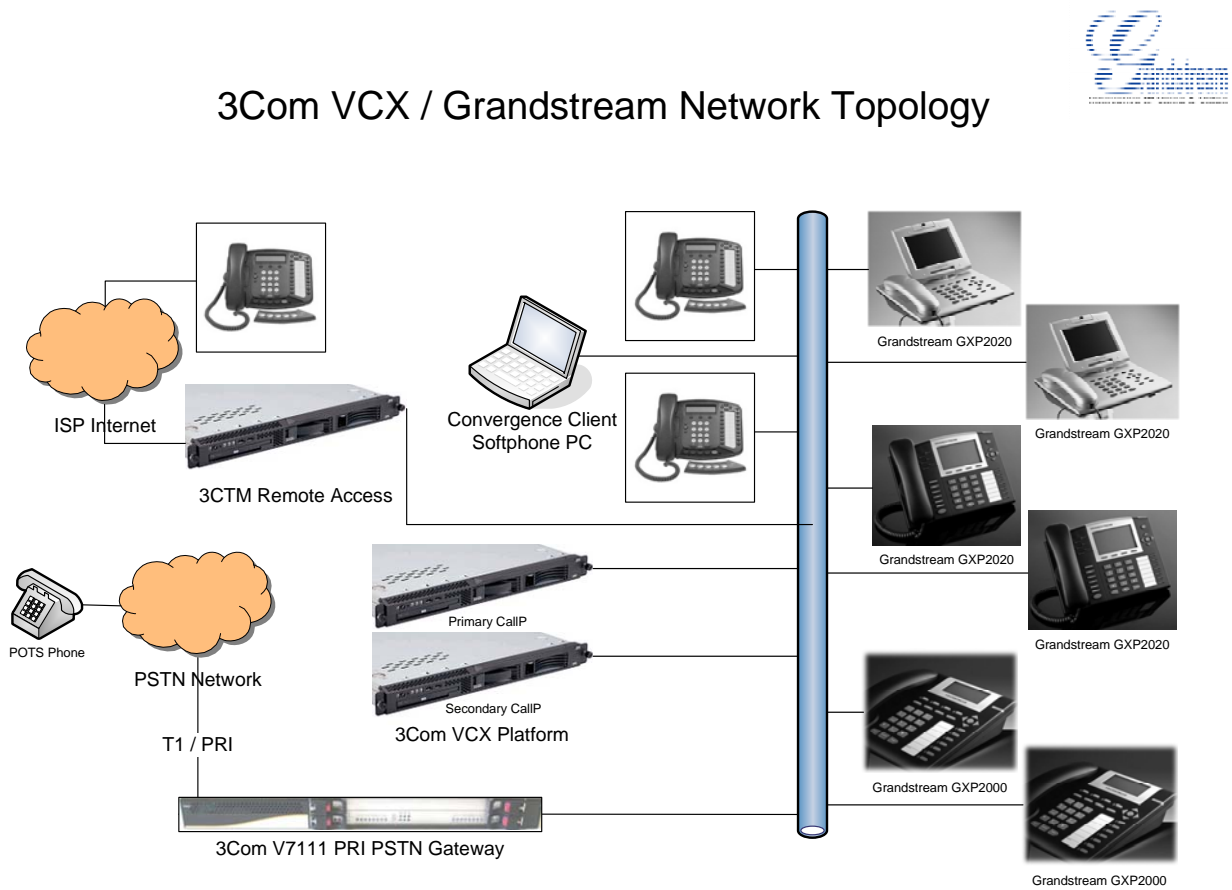


Figure 1: Test Network Topology Diagram

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### ***Basic Setup***

- The setup included 3 GXP Series IP Phones configured with static IPs powered by PoE and/or external Power Supplies.
- User accounts registering to the 3COM VCX system were created for each phone.  
Each phone extension had a unique user created and extension associated to the user. No special settings were needed for these users and extensions.
- PCs on the same network were used to remotely configure the phones as well as capture network traces for analysis and troubleshooting.  
Standard Windows XP O.S. and Linux PCs were available on the network to perform Web Based configuration of the phones, monitoring via a mirror port traffic with Wireshark, and for Convergence Client calls.

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### 3COM Configuration Details

The 5500G switch configuration is presented below. The VLAN id 2 was used as the vlan for the phone testing. The default qos-profile was used. By adding the Grandstream Mac Address ranges into the switch configuration as a rule we can be sure that the Grandstream VoIP traffic is handled appropriately. To do this we added a new rule between rules 7 and 8 of the default switch configuration.

The following data is a subset of the complete switch configuration as tested.

```
#
 poe legacy enable
#
 igmp-snooping enable
#
 multicast routing-enable
#
acl number 3997
 rule 0 permit IP dscp ef
 rule 1 permit TCP destination-port eq www
 rule 2 permit UDP destination-port eq snmp
 rule 3 permit UDP destination-port eq snmptrap
 rule 4 permit IP dscp cs6
 rule 5 permit IP dscp cs7
#
acl number 4999
 rule 0 permit type 8868 ffff
 rule 1 permit source 00e0-bb00-0000 ffff-ff00-0000
 rule 2 permit source 0003-6b00-0000 ffff-ff00-0000
 rule 3 permit source 00e0-7500-0000 ffff-ff00-0000
 rule 4 permit source 00d0-1e00-0000 ffff-ff00-0000
 rule 5 permit source 0001-e300-0000 ffff-ff00-0000
 rule 6 permit source 000f-e200-0000 ffff-ff00-0000
 rule 7 permit source 0006-b900-0000 ffff-ff00-0000
 rule 8 permit source 0008-b200-0000 ffff-ff00-0000 ← Grandstream MAC entry
 rule 9 deny dest 0000-0000-0000 ffff-ffff-ffff ← Original rule 8 edited to be rule 9
#
qos-profile default
 packet-filter inbound link-group 4999 rule 9 ← Changed rule 8 to rule 9
 traffic-priority inbound ip-group 3997 rule 0 cos voice
 traffic-priority inbound ip-group 3997 rule 4 cos network-management
 traffic-priority inbound ip-group 3997 rule 5 cos network-management
 traffic-priority inbound link-group 4999 rule 0 dscp ef cos voice
 traffic-priority inbound link-group 4999 rule 1 dscp ef cos voice
 traffic-priority inbound link-group 4999 rule 2 dscp ef cos voice
 traffic-priority inbound link-group 4999 rule 3 dscp ef cos voice
 traffic-priority inbound link-group 4999 rule 4 dscp ef cos voice
 traffic-priority inbound link-group 4999 rule 5 dscp ef cos voice
 traffic-priority inbound link-group 4999 rule 6 dscp ef cos voice
 traffic-priority inbound link-group 4999 rule 7 dscp ef cos voice
 traffic-priority inbound link-group 4999 rule 8 dscp ef cos voice ← New rule 8
#
#
interface GigabitEthernet1/0/1
 poe enable ← We must turn on Power of Ethernet
 broadcast-suppression pps 3000
 port access vlan 2 ← Port is associated to Vlan id 2
 apply qos-profile default ← QOS profile definition
```

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### **Grandstream Configuration Details**

- Phones were by default configured to DHCP. Using remote HTTP access on the PC, the web configuration screen was accessed at the IP address shown on phone LCD display.



- By default the administrator login password is admin. Upon logging in, **ADVANCED SETTINGS** page is viewed. There are **STATUS**, **BASIC SETTINGS**, and the **ACCOUNT 1-6** pages (GXP2020 has 6 configurable accounts while GXP2000 has 4 accounts)

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### Grandstream Device Configuration

STATUS
BASIC SETTINGS
ADVANCED SETTINGS
ACCOUNT 1
ACCOUNT 2
ACCOUNT 3
ACCOUNT 4
ACCOUNT 5
ACCOUNT 6

**Admin Password:**  (purposely not displayed for security protection)

**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)

**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)

**No Key Entry Timeout:**  (in seconds, default is 4 seconds)

**Use # as Dial Key:**  No  Yes

**local RTP port:**  (1024-65535, default 5004)

**Use random port:**  No  Yes

**keep-alive interval:**  (in seconds, default 20 seconds)

**Use NAT IP**  (if specified, this will be used in SIP/SDP message)

**STUN server:**  (URI or IP:port)

**Firmware Upgrade and Provisioning:** Upgrade Via  TFTP  HTTP

Firmware Server Path:

Config Server Path:

- For testing basic calls this page was kept to default settings. However, for testing VLAN capabilities we later configured Layer 2 QoS settings. Apply the appropriate L2 and L3 QoS values to match the network infrastructure. 3COM Voice products default to using Layer 3 QoS DSCP of 46.

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	Firmware File Prefix: <input type="text"/>
	Firmware File Postfix: <input type="text"/>
	Config File Prefix: <input type="text"/>
	Config File Postfix: <input type="text"/>
	Allow DHCP Option 66 to override server: <input type="radio"/> No <input checked="" type="radio"/> Yes
	Automatic Upgrade: <input checked="" type="radio"/> No <input type="radio"/> Yes, check for upgrade every <input type="text" value="10080"/> minutes (default 7 days)  <input checked="" type="radio"/> Always Check for New Firmware <input type="radio"/> Check New Firmware only when F/W pre/suffix changes <input type="radio"/> Always Skip the Firmware Check
	Authenticate Conf File: <input checked="" type="radio"/> No <input type="radio"/> Yes (cfg file would be authenticated before acceptance if set to Yes)
<i>Phonebook XML Download:</i>	Enable Phonebook XML Download: <input checked="" type="radio"/> No <input type="radio"/> YES, HTTP <input type="radio"/> YES, TFTP
	Phonebook XML Server Path: <input type="text"/>
	Phonebook Download Interval: <input type="text" value="0"/> (0-720)

	Remove Manually-edited entries on Download: <input checked="" type="radio"/> No <input type="radio"/> Yes
<i>Idle Screen XML Download:</i>	Enable Idle Screen XML Download: <input checked="" type="radio"/> No <input type="radio"/> YES, HTTP <input type="radio"/> YES, TFTP
	Idle Screen XML Server Path: <input type="text"/>
<i>DTMF Payload Type:</i>	<input type="text" value="101"/>
<i>Syslog Server:</i>	<input type="text"/>
<i>Syslog Level:</i>	NONE <input type="button" value="v"/>
<i>NTP Server:</i>	<input type="text" value="us.pool.ntp.org"/> (URI or IP address)
	Allow DHCP Option 42 to override NTP server: <input checked="" type="radio"/> No <input type="radio"/> Yes
<i>Distinctive Ring Tone:</i>	Custom ring tone 1, used if incoming caller ID is <input type="text"/>
	Custom ring tone 2, used if incoming caller ID is <input type="text"/>
	Custom ring tone 3, used if incoming caller ID is <input type="text"/>
<i>System Ring Tone:</i>	<input type="text" value="f1=440,f2=480,c=200/400;"/>

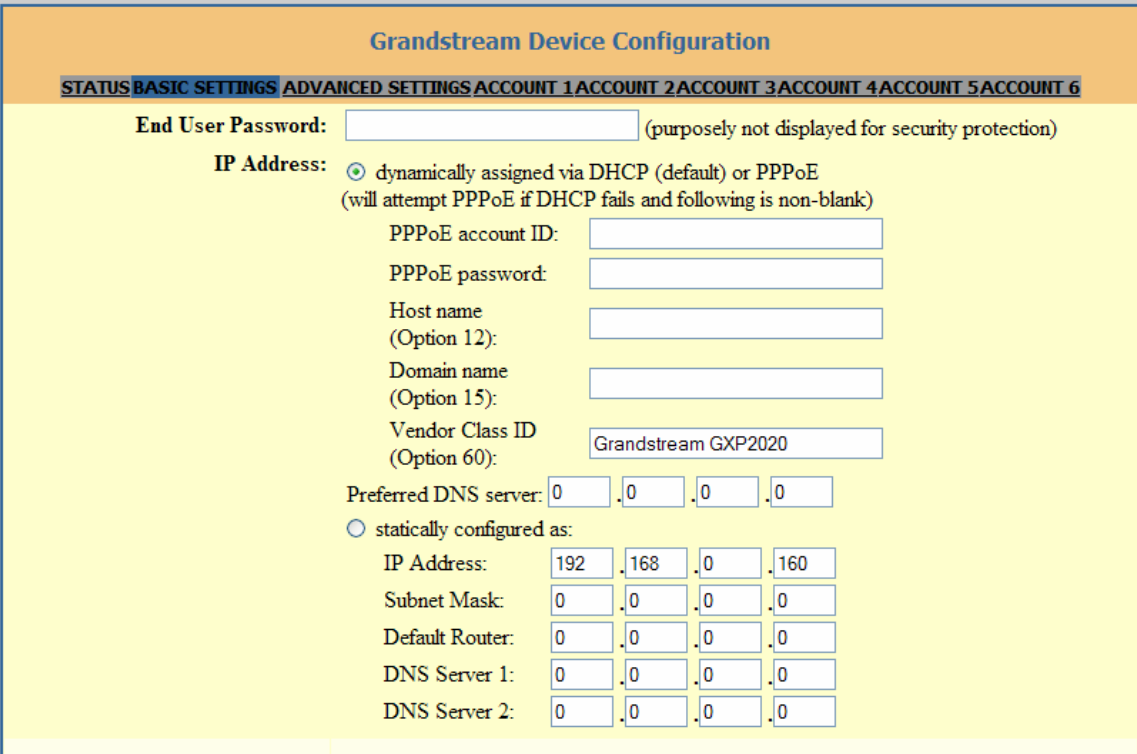
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<i>Call Progress Tones:</i>	Dial Tone	<input type="text" value="f1=350,f2=440;"/>
	Message Waiting	<input type="text" value="f1=350,f2=440,c=10/10;"/>
	Ring Back Tone	<input type="text" value="f1=440,f2=480,c=200/400;"/>
	Call-Waiting Tone	<input type="text" value="f1=440,f2=440,c=25/525;"/>
	Busy Tone	<input type="text" value="f1=480,f2=620,c=50/50;"/>
	Reorder Tone	<input type="text" value="f1=480,f2=620,c=25/25;"/>
Syntax: f1=val, f2=val [, c=on1/off1 [-on2/off2 [-on3/off3] ] ] ; (Frequencies are in Hz and cadence on and off are in 10ms)		
<i>Disable Call-Waiting:</i>	<input checked="" type="radio"/> No <input type="radio"/> Yes	
<i>Disable Call-Waiting Tone:</i>	<input checked="" type="radio"/> No <input type="radio"/> Yes	
<i>Use Quick IP-call mode:</i>	<input checked="" type="radio"/> No <input type="radio"/> Yes	
<i>Lock Keypad Update:</i>	<input checked="" type="radio"/> No <input type="radio"/> Yes (configuration update via keypad is disabled if set to Yes)	
<i>Display Language:</i>	<input type="text" value="English"/>	
<input type="button" value="Update"/> <input type="button" value="Cancel"/> <input type="button" value="Reboot"/>		
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- In order to configure Static IP address click on **BASIC SETTINGS** tab (at the top).



**Grandstream Device Configuration**

**STATUS BASIC SETTINGS ADVANCED SETTINGS ACCOUNT 1 ACCOUNT 2 ACCOUNT 3 ACCOUNT 4 ACCOUNT 5 ACCOUNT 6**

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:

Host name  
(Option 12):

Domain name  
(Option 15):

Vendor Class ID  
(Option 60):

Preferred DNS server:

statically configured as:

IP Address:

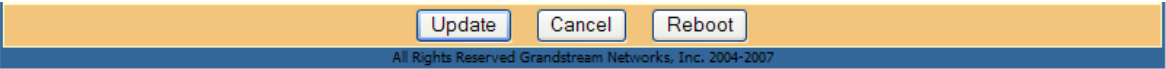
Subnet Mask:

Default Router:

DNS Server 1:

DNS Server 2:

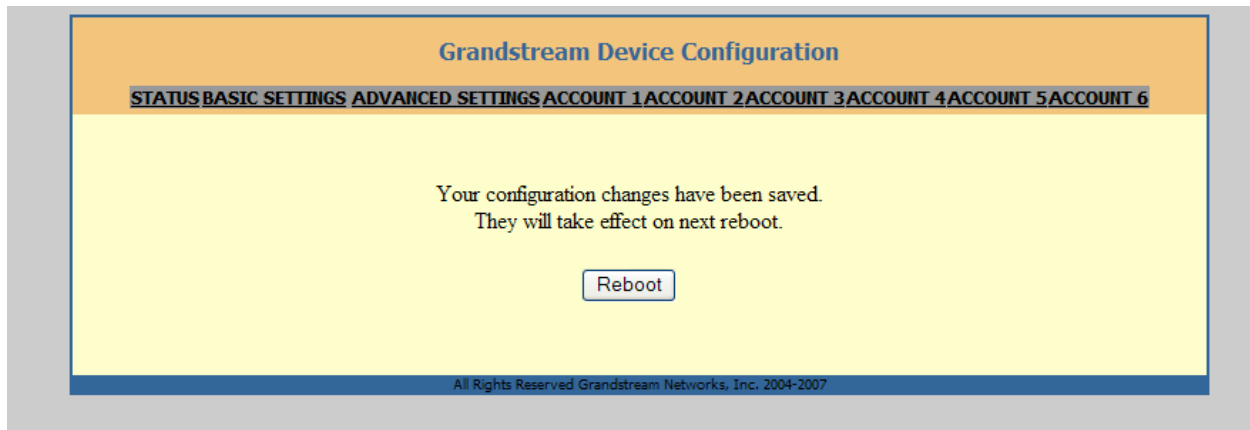
- As you can see by default it is set to DHCP. Click on the ‘**statically configured as**’ radio button and enter the appropriate Static IP information.
- After entering this information, click on **UPDATE** at the bottom to save the changes.



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- Upon clicking **UPDATE**, the following screen will be viewed. Allow for the reboot and log back in.



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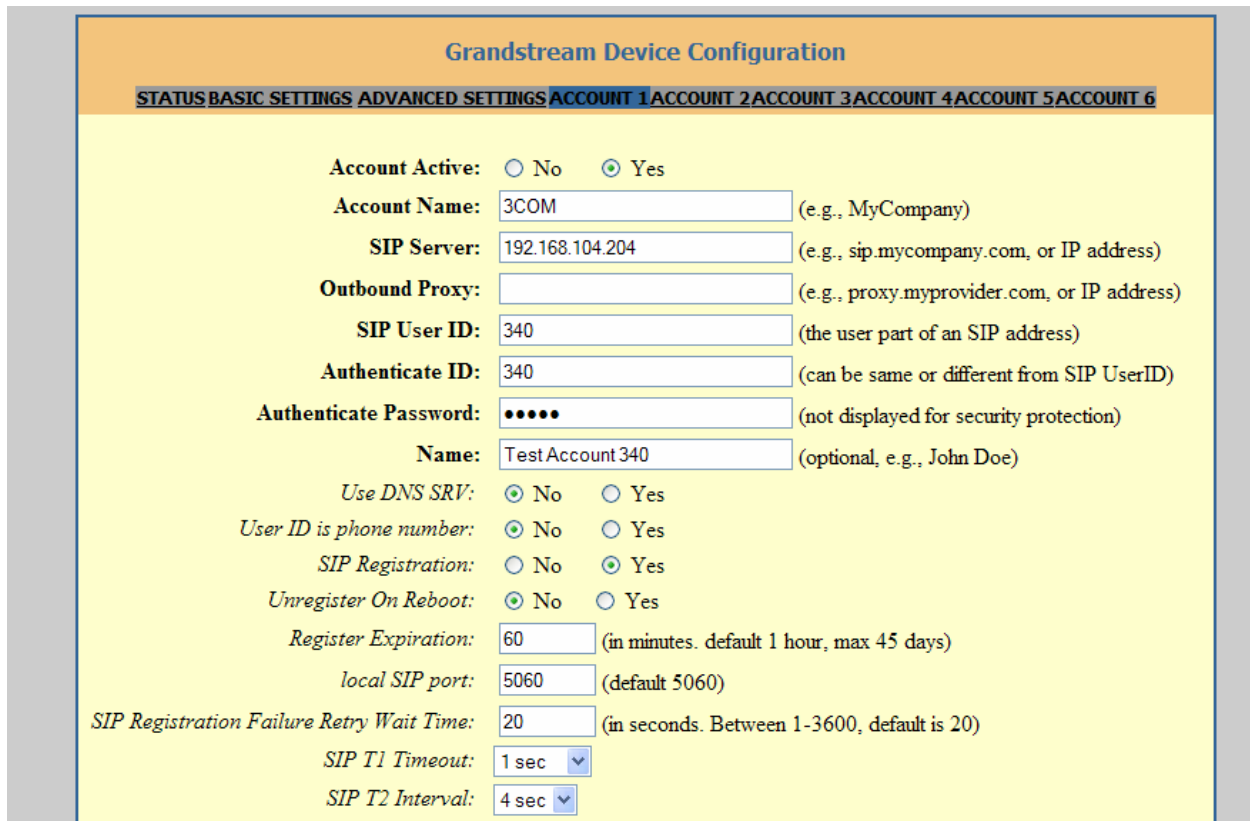
- Log back into the phone and click on **ACCOUNT 1** tab to configure the user account, created on the 3COM VCX.

Enter the appropriate information as shown below:

VCX Primary Call Processor IP: <b>192.168.104.204</b>	← VCX Primary
Phone Extension: <b>340</b>	← Phone Extension
Phone Password: <b>12345</b>	← Phone Password

192.168.104.204 is the IP Address of the VCX Call Processor as given by 3COM. The User account for this phone is 340 and password 12345 (hidden) was entered under **Authenticate Password**.

The **Account Name** and **Name** are optional fields and can be set as per user.



**Grandstream Device Configuration**

STATUS BASIC SETTINGS ADVANCED SETTINGS **ACCOUNT 1** ACCOUNT 2 ACCOUNT 3 ACCOUNT 4 ACCOUNT 5 ACCOUNT 6

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)

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

Authenticate ID:  (can be same or different from SIP UserID)

Authenticate Password:  (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 5060)

SIP Registration Failure Retry Wait Time:  (in seconds. Between 1-3600, default is 20)

SIP T1 Timeout:  ▼

SIP T2 Interval:  ▼

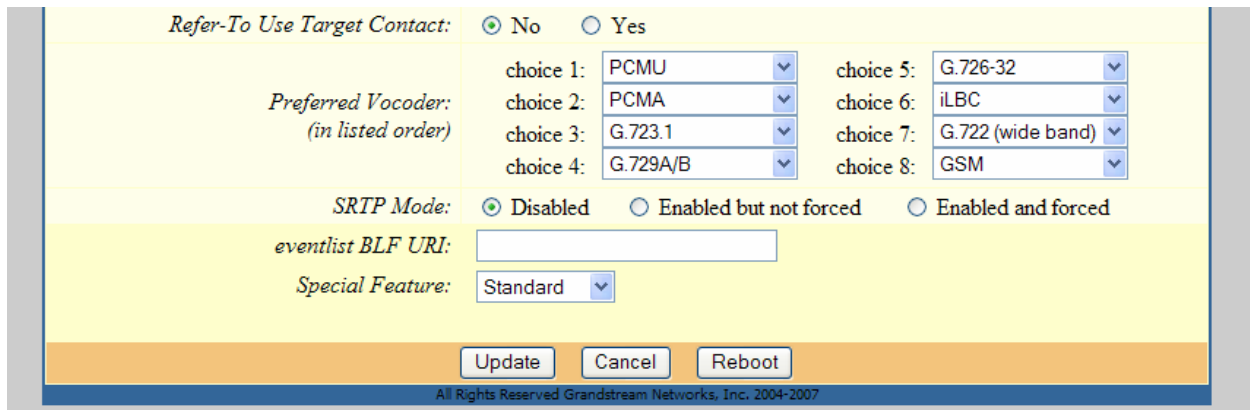
Blue

- Set **Subscribe to MWI** to **Yes** to allow voice mail notifications from the VCX. **Send DTMF** set to **via RTP (RFC2833)**.

<i>SIP Transport:</i>	<input checked="" type="radio"/> UDP <input type="radio"/> TCP
<i>Use RFC3581 Symmetric Routing:</i>	<input checked="" type="radio"/> No <input type="radio"/> Yes
<i>NAT Traversal (STUN):</i>	<input checked="" type="radio"/> No <input type="radio"/> No, but send keep-alive <input type="radio"/> Yes
<i>SUBSCRIBE for MWI:</i>	<input type="radio"/> No <input checked="" type="radio"/> Yes
<i>PUBLISH for Presence:</i>	<input checked="" type="radio"/> No <input type="radio"/> Yes
<i>Proxy-Require:</i>	<input type="text"/>
<i>Voice Mail UserID:</i>	<input type="text"/> (UserID for voice mail system)
<i>Send DTMF:</i>	<input type="checkbox"/> in-audio <input checked="" type="checkbox"/> via RTP (RFC2833) <input type="checkbox"/> via SIP INFO
<i>Early Dial:</i>	<input checked="" type="radio"/> No <input type="radio"/> Yes (use "Yes" only if proxy supports 484 response)
<i>Dial Plan Prefix:</i>	<input type="text"/> (this prefix string is added to each dialed number)
<i>Delayed Call Forward Wait Time:</i>	<input type="text" value="20"/> (Allowed range 1-120, in seconds.)
<i>Enable Call Features:</i>	<input type="radio"/> No <input checked="" type="radio"/> Yes (if yes, call features using star codes will be supported locally)
<i>Call Log:</i>	<input checked="" type="radio"/> Log All Calls <input type="radio"/> Log Incoming/Outgoing only (Missed calls NOT recorded) <input type="radio"/> Disable Call Log
<i>Session Expiration:</i>	<input type="text" value="180"/> (in seconds. default 180 seconds)
<i>Min-SE:</i>	<input type="text" value="90"/> (in seconds. default and minimum 90 seconds)
<i>Caller Request Timer:</i>	<input checked="" type="radio"/> No <input type="radio"/> Yes (Request for timer when making outbound calls)
<i>Callee Request Timer:</i>	<input checked="" type="radio"/> No <input type="radio"/> Yes (When caller supports timer but did not request one)
<i>Force Timer:</i>	<input checked="" type="radio"/> No <input type="radio"/> Yes (Use timer even when remote party does not support)
<i>UAC Specify Refresher:</i>	<input type="radio"/> UAC <input type="radio"/> UAS <input checked="" type="radio"/> Omit (Recommended)

Blue

- You may change the preferred codec used for calls. By default all codecs will be available in the order seen below. You may select all choices to a single codec if you want to force usage of only that codec. Once again, after entering all information, click on **UPDATE** at the bottom to save the changes.



The screenshot shows a configuration panel with a yellow background. At the top, there is a label "Refer-To Use Target Contact:" followed by two radio buttons: "No" (selected) and "Yes". Below this is a section for "Preferred Vocoder: (in listed order)" containing eight dropdown menus labeled "choice 1" through "choice 8". The selected values are: choice 1: PCMU, choice 2: PCMA, choice 3: G.723.1, choice 4: G.729A/B, choice 5: G.726-32, choice 6: iLBC, choice 7: G.722 (wide band), and choice 8: GSM. Below the vocoder choices is the "SRTP Mode:" section with three radio buttons: "Disabled" (selected), "Enabled but not forced", and "Enabled and forced". Underneath is a text input field for "eventlist BLF URI:" which is currently empty. Below that is a dropdown menu for "Special Feature:" with "Standard" selected. At the bottom of the panel are three buttons: "Update", "Cancel", and "Reboot". A small copyright notice "All Rights Reserved Grandstream Networks, Inc. 2004-2007" is visible at the very bottom.

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## Testing Observations

The 3COM VCX and Grandstream GXP Series phones work correctly with each other for the tested call features and functionalities as mentioned on this document.

## Verification Tests

Not all VCX supported features will be available to users through Grandstream phones and vice versa. Some VCX phone features are restricted to 3Com Phone software implementations.

**Note: The support for Primary and Secondary VCX is not available. These phones only maintain registrations with the primary VCX.**

The features that were shown to operate properly are as follows:

- Basic IP Calls
- Basic IP - PSTN gateway calls
- Call Conference
- Call Hold
- Call Transfer (Blind)
- Call Transfer (Attended)
- Call Waiting
- Caller ID
- Do Not Disturb
- Last Number Redial
- Message Waiting Indication
- Missed Call Indicator
- Mute
- Speed Dial
- Calls to/from 3Com 3102 stations
- Convergence Client
- Multi-Way conference calls
- Interoperability with VCX 3CTM

The following tests passed but required using \*XXX codes.

- Automatic Call Back (Camp On)
- Call Forward All
- Call Forward Busy
- Call Forward No answer
- Call Park/Retrieve
- Caller ID Block

## Troubleshooting Tips

Basic troubleshooting technique should suffice for this solution. The integration was very simple and intuitive for a VoIP user/technician.

1. Simply Ping the phone's IP address from a client PC. This will test basic IP connectivity.
2. Double check the phone IP and SIP acct. settings for accuracy. Verify the correct IP network is assigned and default gateway is correct. Ping the default gateway to test for reachability from the client PC. Verify the extension and password are correct.
3. Verify phone registration.
  - a. On the VCX Admin screen navigate to the Phones screen, locate the phone extension associated to the phone and select the Registration link in the left side of the screen. This will report the phone's SIP registration. If its not present the phone is failing to register with the VCX. If it is present the registration is successful.
4. Perform a Wireshark trace of the phone registration.
  - a. Locate an Ethernet hub. Plug the phone line from the wall into the hub, a second Ethernet cable tot eh phone, and a PC running wireshark into the third port.
  - b. Start the capture. And initiate the registration. Use an external power supply to power the phone.
  - c. Compare the registration fields to the registration sample in Appendix A.
  - d. Alternatively a switch mirror group may be provisioned and include the VCX, and Phone port as members of the mirror group. The port the Client PC is connected to will be the monitor port.

Blue

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## Product Support

Product support can be obtained from the respective product suppliers.

### ***3COM product support:***

**Main 3COM Support link:**

[http://www.3com.com/products/en\\_US/support/index.html](http://www.3com.com/products/en_US/support/index.html)

**3COM Product Specific Link**

[http://www.3com.com/products/en\\_US/detail.jsp?pathtype=purchase&tab=features&sku=WEBBN GVCXV7000](http://www.3com.com/products/en_US/detail.jsp?pathtype=purchase&tab=features&sku=WEBBN GVCXV7000)

**Asia Pacific**

Telephone: +65 6543 6645

Fax: +65 6543 6518

E-mail: [ap\\_service@3com.com](mailto:ap_service@3com.com)

**Europe, Middle East and Africa**

Telephone: +44 (0)1442 435529 (Option 4)

Fax : +44 (0)1442 435811

E-mail: [focalpoint\\_services@3com.com](mailto:focalpoint_services@3com.com)

**North America and Latin America**

Telephone: 866-326-6222 (Option 3)

Fax : 408-326-7140

E-mail: [ecso\\_contracts@3com.com](mailto:ecso_contracts@3com.com)

### ***Grandstream Product Support:***

<http://www.grandstream.com/gxp2020.html>

<http://www.grandstream.com/gxp2000.html>

**Grandstream Networks**

Support Web page: <http://www.grandstream.com/customersupport.html>

**Support:**

617-566-9300 x2

[support@grandstream.com](mailto:support@grandstream.com)



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## **Conclusion**

The GXP series is the choice for business customers looking for an affordable, simple and feature rich VoIP phone when using the 3Com VCX platform. For system integrators, the Grandstream SIP phone portfolio will help maximize margins and increase the customer value of the 3Com VCX / Grandstream solution. Choose the 3Com VCX platform and Grandstream SIP phones for the most cost-effective VoIP solution for your customers.

Blue

## Appendix A: Phone Registration

```
Frame 1 (565 bytes on wire, 565 bytes captured)
  Arrival Time: May 30, 2007 10:51:19.697352000
  [Time delta from previous packet: 0.000000000 seconds]
  [Time since reference or first frame: 0.000000000 seconds]
  Frame Number: 1
  Packet Length: 565 bytes
  Capture Length: 565 bytes
  [Frame is marked: False]
  [Protocols in frame: eth:ip:udp:sip]
  [Coloring Rule Name: UDP]
  [Coloring Rule String: udp]
Ethernet II, Src: Grandstr_0e:e1:17 (00:0b:82:0e:e1:17), Dst: Ibm_22:1a:1c (00:11:25:22:1a:1c)
  Destination: Ibm_22:1a:1c (00:11:25:22:1a:1c)
    Address: Ibm_22:1a:1c (00:11:25:22:1a:1c)
      ....0... = IG bit: Individual address (unicast)
      ....0... = LG bit: Globally unique address (factory default)
  Source: Grandstr_0e:e1:17 (00:0b:82:0e:e1:17)
    Address: Grandstr_0e:e1:17 (00:0b:82:0e:e1:17)
      ....0... = IG bit: Individual address (unicast)
      ....0... = LG bit: Globally unique address (factory default)
  Type: IP (0x0800)
Internet Protocol, Src: 192.168.104.246 (192.168.104.246), Dst: 192.168.104.204 (192.168.104.204)
  Version: 4
  Header length: 20 bytes
  Differentiated Services Field: 0xc0 (DSCP 0x30: Class Selector 6; ECN: 0x00)
    1100 00.. = Differentiated Services Codepoint: Class Selector 6 (0x30)
    ....0.. = ECN-Capable Transport (ECT): 0
    ....0.. = ECN-CE: 0
  Total Length: 551
  Identification: 0x0000 (0)
  Flags: 0x00
    0... = Reserved bit: Not set
    .0.. = Don't fragment: Not set
    ..0. = More fragments: Not set
  Fragment offset: 0
  Time to live: 255
  Protocol: UDP (0x11)
  Header checksum: 0x65f2 [correct]
    [Good: True]
    [Bad : False]
  Source: 192.168.104.246 (192.168.104.246)
  Destination: 192.168.104.204 (192.168.104.204)
User Datagram Protocol, Src Port: 5060 (5060), Dst Port: 5060 (5060)
  Source port: 5060 (5060)
  Destination port: 5060 (5060)
  Length: 531
  Checksum: 0x394c [correct]
Session Initiation Protocol
  Request-Line: REGISTER sip:192.168.104.204 SIP/2.0
    Method: REGISTER
    [Resent Packet: False]
  Message Header
    Via: SIP/2.0/UDP 192.168.104.246;branch=z9hG4bKa2e550de098cde1e
      Transport: UDP
      Sent-by Address: 192.168.104.246
      Sent-by port: 5060
      Branch: z9hG4bKa2e550de098cde1e
    From: "3COM 341" <sip:341@192.168.104.204>;tag=ca166cec5a6f785a
      SIP Display info: "3COM 341"
      SIP from address: sip:341@192.168.104.204
      SIP tag: ca166cec5a6f785a
```

Blue

```
To: <sip:341@192.168.104.204>
  SIP to address: sip:341@192.168.104.204
Contact: <sip:341@192.168.104.246:5060;transport=udp>
  Contact Binding: <sip:341@192.168.104.246:5060;transport=udp>
    URI: <sip:341@192.168.104.246:5060;transport=udp>
    SIP contact address: sip:341@192.168.104.246:5060
Supported: path
Call-ID: b81cd92cb36fd519@192.168.104.246
CSeq: 10001 REGISTER
  Sequence Number: 10001
  Method: REGISTER
Expires: 3600
User-Agent: Grandstream GXP2020 1.1.4.6
Max-Forwards: 70
Allow: INVITE,ACK,CANCEL,BYE,NOTIFY,REFER,OPTIONS,INFO,SUBSCRIBE,UPDATE,PRACK,MESSAGE
Content-Length: 0
```

**Frame 2 (325 bytes on wire, 325 bytes captured)**

```
Arrival Time: May 30, 2007 10:51:19.698256000
[Time delta from previous packet: 0.000904000 seconds]
[Time since reference or first frame: 0.000904000 seconds]
Frame Number: 2
Packet Length: 325 bytes
Capture Length: 325 bytes
[Frame is marked: False]
[Protocols in frame: eth:ip:udp:sip]
[Coloring Rule Name: UDP]
[Coloring Rule String: udp]
Ethernet II, Src: Ibm_22:1a:1c (00:11:25:22:1a:1c), Dst: Grandstr_0e:e1:17 (00:0b:82:0e:e1:17)
  Destination: Grandstr_0e:e1:17 (00:0b:82:0e:e1:17)
    Address: Grandstr_0e:e1:17 (00:0b:82:0e:e1:17)
    ....0... = IG bit: Individual address (unicast)
    ....0... = LG bit: Globally unique address (factory default)
  Source: Ibm_22:1a:1c (00:11:25:22:1a:1c)
    Address: Ibm_22:1a:1c (00:11:25:22:1a:1c)
    ....0... = IG bit: Individual address (unicast)
    ....0... = LG bit: Globally unique address (factory default)
  Type: IP (0x0800)
Internet Protocol, Src: 192.168.104.204 (192.168.104.204), Dst: 192.168.104.246 (192.168.104.246)
  Version: 4
  Header length: 20 bytes
  Differentiated Services Field: 0xb8 (DSCP 0x2e: Expedited Forwarding; ECN: 0x00)
    1011 10.. = Differentiated Services Codepoint: Expedited Forwarding (0x2e)
    ....0.. = ECN-Capable Transport (ECT): 0
    ....0.. = ECN-CE: 0
  Total Length: 311
  Identification: 0x0000 (0)
  Flags: 0x04 (Don't Fragment)
    0... = Reserved bit: Not set
    .1.. = Don't fragment: Set
    ..0. = More fragments: Not set
  Fragment offset: 0
  Time to live: 64
  Protocol: UDP (0x11)
  Header checksum: 0xe5ea [correct]
    [Good: True]
    [Bad: False]
  Source: 192.168.104.204 (192.168.104.204)
  Destination: 192.168.104.246 (192.168.104.246)
User Datagram Protocol, Src Port: 5060 (5060), Dst Port: 5060 (5060)
  Source port: 5060 (5060)
  Destination port: 5060 (5060)
  Length: 291
  Checksum: 0xe7df [correct]
```

Blue

```

Session Initiation Protocol
  Status-Line: SIP/2.0 100 Trying
    Status-Code: 100
    [Resent Packet: False]
  Message Header
    v: SIP/2.0/UDP 192.168.104.246:5060;branch=z9hG4bKa2e550de098cde1e
      Transport: UDP
      Sent-by Address: 192.168.104.246
      Sent-by port: 5060
      Branch: z9hG4bKa2e550de098cde1e
    f: "3COM 341"<sip:341@192.168.104.204>;tag=cal66cec5a6f785a
      SIP Display info: "3COM 341"
      SIP from address: sip:341@192.168.104.204
      SIP tag: cal66cec5a6f785a
    t: <sip:341@192.168.104.204>
      SIP to address: sip:341@192.168.104.204
    i: b81cd92cb36fd519@192.168.104.246
    Cseq: 10001 REGISTER
      Sequence Number: 10001
      Method: REGISTER
    Date: Wed, 30 May 2007 13:50:07 GMT
    Content-Length: 0

Frame 3 (657 bytes on wire, 657 bytes captured)
  Arrival Time: May 30, 2007 10:51:19.699148000
  [Time delta from previous packet: 0.000892000 seconds]
  [Time since reference or first frame: 0.001796000 seconds]
  Frame Number: 3
  Packet Length: 657 bytes
  Capture Length: 657 bytes
  [Frame is marked: False]
  [Protocols in frame: eth:ip:udp:sip]
  [Coloring Rule Name: UDP]
  [Coloring Rule String: udp]
Ethernet II, Src: Ibm_22:1a:1c (00:11:25:22:1a:1c), Dst: Grandstr_0e:e1:17 (00:0b:82:0e:e1:17)
  Destination: Grandstr_0e:e1:17 (00:0b:82:0e:e1:17)
    Address: Grandstr_0e:e1:17 (00:0b:82:0e:e1:17)
      ....0... = IG bit: Individual address (unicast)
      ...0... = LG bit: Globally unique address (factory default)
  Source: Ibm_22:1a:1c (00:11:25:22:1a:1c)
    Address: Ibm_22:1a:1c (00:11:25:22:1a:1c)
      ....0... = IG bit: Individual address (unicast)
      ...0... = LG bit: Globally unique address (factory default)
  Type: IP (0x0800)
Internet Protocol, Src: 192.168.104.204 (192.168.104.204), Dst: 192.168.104.246 (192.168.104.246)
  Version: 4
  Header length: 20 bytes
  Differentiated Services Field: 0xb8 (DSCP 0x2e: Expedited Forwarding; ECN: 0x00)
    1011 10.. = Differentiated Services Codepoint: Expedited Forwarding (0x2e)
      ...0.. = ECN-Capable Transport (ECT): 0
      ....0 = ECN-CE: 0
  Total Length: 643
  Identification: 0x0000 (0)
  Flags: 0x04 (Don't Fragment)
    0... = Reserved bit: Not set
    1.. = Don't fragment: Set
    ..0 = More fragments: Not set
  Fragment offset: 0
  Time to live: 64
  Protocol: UDP (0x11)
  Header checksum: 0xe49e [correct]
    [Good: True]
    [Bad : False]
  Source: 192.168.104.204 (192.168.104.204)

```

Blue

```

Destination: 192.168.104.246 (192.168.104.246)
User Datagram Protocol, Src Port: 5060 (5060), Dst Port: 5060 (5060)
Source port: 5060 (5060)
Destination port: 5060 (5060)
Length: 623
Checksum: 0x11ed [correct]
Session Initiation Protocol
Status-Line: SIP/2.0 401 Unauthorized
  Status-Code: 401
  [Resent Packet: False]
Message Header
  v: SIP/2.0/UDP 192.168.104.246:5060;branch=z9hG4bKa2e550de098cde1e
    Transport: UDP
    Sent-by Address: 192.168.104.246
    Sent-by port: 5060
    Branch: z9hG4bKa2e550de098cde1e
  f: "3COM 341"<sip:341@192.168.104.204>;tag=cal66cec5a6f785a
    SIP Display info: "3COM 341"
    SIP from address: sip:341@192.168.104.204
    SIP tag: cal66cec5a6f785a
  t: <sip:341@192.168.104.204>;tag=820f614
    SIP to address: sip:341@192.168.104.204
    SIP tag: 820f614
  i: b81cd92cb36fd519@192.168.104.246
Cseq: 10001 REGISTER
  Sequence Number: 10001
  Method: REGISTER
Date: Wed, 30 May 2007 13:50:07 GMT
Allow: INVITE,ACK,BYE,CANCEL,REFER,SUBSCRIBE,NOTIFY,UPDATE,OPTIONS,MESSAGE,FEATURE
k: path
Expires: 3600
User-Agent: 3Com VCX 7210 IP CallProcessor/v7.1.42
WWW-Authenticate: Digest realm="3Com", domain="3Com",
nonce="aLaLaSaKaPaNaNaKaKaRaUaLaTaMaIaLaQaSaIaLaKaOaIaMaOaQaUaPaKaQaK", stale=FALSE,
algorithm=MD5
  Authentication Scheme: Digest
  Realm: "3Com"
  Authentication Domain: "3Com"
  Nonce Value: "aLaLaSaKaPaNaNaKaKaRaUaLaTaMaIaLaQaSaIaLaKaOaIaMaOaQaUaPaKaQaK"
  Stale Flag: FALSE
  Algorithm: MD5
Content-Length: 0

```

**Frame 4 (776 bytes on wire, 776 bytes captured)**

```

Arrival Time: May 30, 2007 10:51:19.741527000
[Time delta from previous packet: 0.042379000 seconds]
[Time since reference or first frame: 0.044175000 seconds]
Frame Number: 4
Packet Length: 776 bytes
Capture Length: 776 bytes
[Frame is marked: False]
[Protocols in frame: eth:ip:udp:sip]
[Coloring Rule Name: UDP]
[Coloring Rule String: udp]
Ethernet II, Src: Grandstr_0e:e1:17 (00:0b:82:0e:e1:17), Dst: Ibm_22:1a:1c (00:11:25:22:1a:1c)
Destination: Ibm_22:1a:1c (00:11:25:22:1a:1c)
Address: Ibm_22:1a:1c (00:11:25:22:1a:1c)
.... 0 .... = IG bit: Individual address (unicast)
.... 0 .... = LG bit: Globally unique address (factory default)
Source: Grandstr_0e:e1:17 (00:0b:82:0e:e1:17)
Address: Grandstr_0e:e1:17 (00:0b:82:0e:e1:17)
.... 0 .... = IG bit: Individual address (unicast)
.... 0 .... = LG bit: Globally unique address (factory default)
Type: IP (0x0800)

```

## Blue

```
Internet Protocol, Src: 192.168.104.246 (192.168.104.246), Dst: 192.168.104.204 (192.168.104.204)
Version: 4
Header length: 20 bytes
Differentiated Services Field: 0xc0 (DSCP 0x30: Class Selector 6; ECN: 0x00)
    1100 00.. = Differentiated Services Codepoint: Class Selector 6 (0x30)
    .... ..0. = ECN-Capable Transport (ECT): 0
    .... ...0 = ECN-CE: 0
Total Length: 762
Identification: 0x0001 (1)
Flags: 0x00
    0... = Reserved bit: Not set
    .0.. = Don't fragment: Not set
    ..0. = More fragments: Not set
Fragment offset: 0
Time to live: 255
Protocol: UDP (0x11)
Header checksum: 0x651e [correct]
    [Good: True]
    [Bad : False]
Source: 192.168.104.246 (192.168.104.246)
Destination: 192.168.104.204 (192.168.104.204)
User Datagram Protocol, Src Port: 5060 (5060), Dst Port: 5060 (5060)
Source port: 5060 (5060)
Destination port: 5060 (5060)
Length: 742
Checksum: 0xb941 [correct]
Session Initiation Protocol
Request-Line: REGISTER sip:192.168.104.204 SIP/2.0
Method: REGISTER
[Resent Packet: False]
Message Header
Via: SIP/2.0/UDP 192.168.104.246:5060;branch=z9hG4bKe0b2f10bbb83f87a
Transport: UDP
Sent-by Address: 192.168.104.246
Sent-by port: 5060
Branch: z9hG4bKe0b2f10bbb83f87a
From: "3COM 341" <sip:341@192.168.104.204>;tag=cal66cec5a6f785a
SIP Display info: "3COM 341"
SIP from address: sip:341@192.168.104.204
SIP tag: cal66cec5a6f785a
To: <sip:341@192.168.104.204>
SIP to address: sip:341@192.168.104.204
Contact: <sip:341@192.168.104.246:5060;transport=udp>
Contact Binding: <sip:341@192.168.104.246:5060;transport=udp>
URI: <sip:341@192.168.104.246:5060;transport=udp>
SIP contact address: sip:341@192.168.104.246:5060
Supported: path
Authorization: Digest username="341", realm="3Com", algorithm=MD5,
uri="sip:192.168.104.204",
nonce="aLaLaSaKaPaNaNaKaKaRaUaLaTaMaIaLaQaSaIaLaKaOaIaMaOaQaUaPaKaQaK",
response="2f2391ee4abaf89dd3b29f3832979227"
Authentication Scheme: Digest
Username: "341"
Realm: "3Com"
Algorithm: MD5
Authentication URI: "sip:192.168.104.204"
Nonce Value: "aLaLaSaKaPaNaNaKaKaRaUaLaTaMaIaLaQaSaIaLaKaOaIaMaOaQaUaPaKaQaK"
Digest Authentication Response: "2f2391ee4abaf89dd3b29f3832979227"
Call-ID: b81cd92cb36fd519@192.168.104.246
CSeq: 10002 REGISTER
Sequence Number: 10002
Method: REGISTER
Expires: 3600
User-Agent: Grandstream GXP2020 1.1.4.6
Max-Forwards: 70
```

Blue

Allow: INVITE,ACK,CANCEL,BYE,NOTIFY,REFER,OPTIONS,INFO,SUBSCRIBE,UPDATE,PRACK,MESSAGE  
Content-Length: 0

**Frame 5 (325 bytes on wire, 325 bytes captured)**

Arrival Time: May 30, 2007 10:51:19.742283000  
[Time delta from previous packet: 0.000756000 seconds]  
[Time since reference or first frame: 0.044931000 seconds]  
Frame Number: 5  
Packet Length: 325 bytes  
Capture Length: 325 bytes  
[Frame is marked: False]  
[Protocols in frame: eth:ip:udp:sip]  
[Coloring Rule Name: UDP]  
[Coloring Rule String: udp]

Ethernet II, Src: Ibm\_22:1a:1c (00:11:25:22:1a:1c), Dst: Grandstr\_0e:e1:17 (00:0b:82:0e:e1:17)

Destination: Grandstr\_0e:e1:17 (00:0b:82:0e:e1:17)  
Address: Grandstr\_0e:e1:17 (00:0b:82:0e:e1:17)  
.... 0... = IG bit: Individual address (unicast)  
.... 0... = LG bit: Globally unique address (factory default)

Source: Ibm\_22:1a:1c (00:11:25:22:1a:1c)  
Address: Ibm\_22:1a:1c (00:11:25:22:1a:1c)  
.... 0... = IG bit: Individual address (unicast)  
.... 0... = LG bit: Globally unique address (factory default)

Type: IP (0x0800)

Internet Protocol, Src: 192.168.104.204 (192.168.104.204), Dst: 192.168.104.246 (192.168.104.246)

Version: 4  
Header length: 20 bytes  
Differentiated Services Field: 0xb8 (DSCP 0x2e: Expedited Forwarding; ECN: 0x00)  
1011 10.. = Differentiated Services Codepoint: Expedited Forwarding (0x2e)  
.... 0.. = ECN-Capable Transport (ECT): 0  
.... 0.. = ECN-CE: 0

Total Length: 311  
Identification: 0x0000 (0)  
Flags: 0x04 (Don't Fragment)  
0... = Reserved bit: Not set  
.1.. = Don't fragment: Set  
..0. = More fragments: Not set

Fragment offset: 0  
Time to live: 64  
Protocol: UDP (0x11)  
Header checksum: 0xe5ea [correct]  
[Good: True]  
[Bad : False]

Source: 192.168.104.204 (192.168.104.204)  
Destination: 192.168.104.246 (192.168.104.246)

User Datagram Protocol, Src Port: 5060 (5060), Dst Port: 5060 (5060)

Source port: 5060 (5060)  
Destination port: 5060 (5060)  
Length: 291

Checksum: 0xaf1e [correct]

Session Initiation Protocol

Status-Line: SIP/2.0 100 Trying  
Status-Code: 100  
[Resent Packet: False]

Message Header

v: SIP/2.0/UDP 192.168.104.246:5060;branch=z9hG4bKe0b2f10bbb83f87a  
Transport: UDP  
Sent-by Address: 192.168.104.246  
Sent-by port: 5060  
Branch: z9hG4bKe0b2f10bbb83f87a  
f: "3COM 341"<sip:341@192.168.104.204>;tag=ca166cec5a6f785a  
SIP Display info: "3COM 341"  
SIP from address: sip:341@192.168.104.204  
SIP tag: ca166cec5a6f785a

Blue

```
t: <sip:341@192.168.104.204>
  SIP to address: sip:341@192.168.104.204
i: b81cd92cb36fd519@192.168.104.246
Cseq: 10002 REGISTER
  Sequence Number: 10002
  Method: REGISTER
Date: Wed, 30 May 2007 13:50:07 GMT
Content-Length: 0
```

**Frame 6 (528 bytes on wire, 528 bytes captured)**

```
Arrival Time: May 30, 2007 10:51:19.793395000
[Time delta from previous packet: 0.051112000 seconds]
[Time since reference or first frame: 0.096043000 seconds]
Frame Number: 6
Packet Length: 528 bytes
Capture Length: 528 bytes
[Frame is marked: False]
[Protocols in frame: eth:ip:udp:sip]
[Coloring Rule Name: UDP]
[Coloring Rule String: udp]
Ethernet II, Src: Ibm_22:1a:1c (00:11:25:22:1a:1c), Dst: Grandstr_0e:e1:17 (00:0b:82:0e:e1:17)
  Destination: Grandstr_0e:e1:17 (00:0b:82:0e:e1:17)
  Address: Grandstr_0e:e1:17 (00:0b:82:0e:e1:17)
  ....0.0. .... = IG bit: Individual address (unicast)
  ....0. .... = LG bit: Globally unique address (factory default)
  Source: Ibm_22:1a:1c (00:11:25:22:1a:1c)
  Address: Ibm_22:1a:1c (00:11:25:22:1a:1c)
  ....0. .... = IG bit: Individual address (unicast)
  ....0. .... = LG bit: Globally unique address (factory default)
Type: IP (0x0800)
Internet Protocol, Src: 192.168.104.204 (192.168.104.204), Dst: 192.168.104.246 (192.168.104.246)
Version: 4
Header length: 20 bytes
Differentiated Services Field: 0xb8 (DSCP 0x2e: Expedited Forwarding; ECN: 0x00)
  1011 10.. = Differentiated Services Codepoint: Expedited Forwarding (0x2e)
  ....0. = ECN-Capable Transport (ECT): 0
  ....0. = ECN-CE: 0
Total Length: 514
Identification: 0x0000 (0)
Flags: 0x04 (Don't Fragment)
  0... = Reserved bit: Not set
  .1.. = Don't fragment: Set
  ..0. = More fragments: Not set
Fragment offset: 0
Time to live: 64
Protocol: UDP (0x11)
Header checksum: 0xe51f [correct]
  [Good: True]
  [Bad : False]
Source: 192.168.104.204 (192.168.104.204)
Destination: 192.168.104.246 (192.168.104.246)
User Datagram Protocol, Src Port: 5060 (5060), Dst Port: 5060 (5060)
Source port: 5060 (5060)
Destination port: 5060 (5060)
Length: 494
Checksum: 0x1391 [correct]
Session Initiation Protocol
Status-Line: SIP/2.0 200 OK
Status-Code: 200
[Resent Packet: False]
Message Header
v: SIP/2.0/UDP 192.168.104.246:5060;branch=z9hG4bKe0b2f10bbb83f87a
Transport: UDP
Sent-by Address: 192.168.104.246
```



## Blue

```
Sent-by port: 5060
Branch: z9hG4bKe0b2f10bbb83f87a
f: "3COM 341"<sip:341@192.168.104.204>;tag=cal66cec5a6f785a
SIP Display info: "3COM 341"
SIP from address: sip:341@192.168.104.204
SIP tag: cal66cec5a6f785a
t: <sip:341@192.168.104.204>;tag=821a584
SIP to address: sip:341@192.168.104.204
SIP tag: 821a584
i: b81cd92cb36fd519@192.168.104.246
Cseq: 10002 REGISTER
Sequence Number: 10002
Method: REGISTER
Date: Wed, 30 May 2007 13:50:07 GMT
m: <sip:341@192.168.104.246:5060>
Contact Binding: <sip:341@192.168.104.246:5060>
URI: <sip:341@192.168.104.246:5060>
SIP contact address: sip:341@192.168.104.246:5060
Allow: INVITE,ACK,BYE,CANCEL,REFER,SUBSCRIBE,NOTIFY,UPDATE,OPTIONS,MESSAGE,FEATURE
k: path
Expires: 3600
User-Agent: 3Com VCX 7210 IP CallProcessor/v7.1.42
Content-Length: 0
```

### Frame 7 (650 bytes on wire, 650 bytes captured)

```
Arrival Time: May 30, 2007 10:51:19.794411000
[Time delta from previous packet: 0.001016000 seconds]
[Time since reference or first frame: 0.097059000 seconds]
Frame Number: 7
Packet Length: 650 bytes
Capture Length: 650 bytes
[Frame is marked: False]
[Protocols in frame: eth:ip:udp:sip]
[Coloring Rule Name: UDP]
[Coloring Rule String: udp]
Ethernet II, Src: Ibm_22:1a:1c (00:11:25:22:1a:1c), Dst: Grandstr_0e:e1:17 (00:0b:82:0e:e1:17)
Destination: Grandstr_0e:e1:17 (00:0b:82:0e:e1:17)
Address: Grandstr_0e:e1:17 (00:0b:82:0e:e1:17)
.... 0. .... = IG bit: Individual address (unicast)
.... 0. .... = LG bit: Globally unique address (factory default)
Source: Ibm_22:1a:1c (00:11:25:22:1a:1c)
Address: Ibm_22:1a:1c (00:11:25:22:1a:1c)
.... 0. .... = IG bit: Individual address (unicast)
.... 0. .... = LG bit: Globally unique address (factory default)
Type: IP (0x0800)
Internet Protocol, Src: 192.168.104.204 (192.168.104.204), Dst: 192.168.104.246 (192.168.104.246)
Version: 4
Header length: 20 bytes
Differentiated Services Field: 0xb8 (DSCP 0x2e: Expedited Forwarding; ECN: 0x00)
1011 10.. = Differentiated Services Codepoint: Expedited Forwarding (0x2e)
.... 0. = ECN-Capable Transport (ECT): 0
.... 0. = ECN-CE: 0
Total Length: 636
Identification: 0x0000 (0)
Flags: 0x04 (Don't Fragment)
0... = Reserved bit: Not set
.1.. = Don't fragment: Set
..0. = More fragments: Not set
Fragment offset: 0
Time to live: 64
Protocol: UDP (0x11)
Header checksum: 0xe4a5 [correct]
[Good: True]
[Bad : False]
```

Blue

```

Source: 192.168.104.204 (192.168.104.204)
Destination: 192.168.104.246 (192.168.104.246)
User Datagram Protocol, Src Port: 5060 (5060), Dst Port: 5060 (5060)
Source port: 5060 (5060)
Destination port: 5060 (5060)
Length: 616
Checksum: 0x5be0 [correct]
Session Initiation Protocol
Request-Line: NOTIFY sip:341@192.168.104.246:5060 SIP/2.0
Method: NOTIFY
[Resent Packet: False]
Message Header
v: SIP/2.0/UDP 192.168.104.204;branch=z9hG4bK8011f557-220d-dc11-887a-b53b2d69a9ca
Transport: UDP
Sent-by Address: 192.168.104.204
Branch: z9hG4bK8011f557-220d-dc11-887a-b53b2d69a9ca
f: <sip:192.168.104.204>;tag=b3e667d4
SIP from address: sip:192.168.104.204
SIP tag: b3e667d4
t: <sip:341@192.168.104.204>
SIP to address: sip:341@192.168.104.204
i: 80b690e7-750c-dc11-a771-d53e931519da
Cseq: 51 NOTIFY
Sequence Number: 51
Method: NOTIFY
Timestamp: 1180533007
Date: Wed, 30 May 2007 13:50:07 GMT
Max-Forwards: 70
m: <sip:3ComCallProcessor@192.168.104.204>
Contact Binding: <sip:3ComCallProcessor@192.168.104.204>
URI: <sip:3ComCallProcessor@192.168.104.204>
SIP contact address: sip:3ComCallProcessor@192.168.104.204
Event: message-summary
User-Agent: 3Com VCX 7210 IP CallProcessor/v7.1.42
Subscription-State: active;expires=3600
c: application/simple-message-summary
Content-Length: 66
Message body
Messages-Waiting: no\r\n
Message-Account: sip:341@192.168.104.204\r\n
\r\n
Frame 8 (649 bytes on wire, 649 bytes captured)
Arrival Time: May 30, 2007 10:51:19.821276000
[Time delta from previous packet: 0.026865000 seconds]
[Time since reference or first frame: 0.123924000 seconds]
Frame Number: 8
Packet Length: 649 bytes
Capture Length: 649 bytes
[Frame is marked: False]
[Protocols in frame: eth:ip:udp:sip]
[Coloring Rule Name: UDP]
[Coloring Rule String: udp]
Ethernet II, Src: Grandstr_0e:e1:17 (00:0b:82:0e:e1:17), Dst: Ibm_22:1a:1c (00:11:25:22:1a:1c)
Destination: Ibm_22:1a:1c (00:11:25:22:1a:1c)
Address: Ibm_22:1a:1c (00:11:25:22:1a:1c)
.... 0. .... = IG bit: Individual address (unicast)
.... 0. .... = LG bit: Globally unique address (factory default)
Source: Grandstr_0e:e1:17 (00:0b:82:0e:e1:17)
Address: Grandstr_0e:e1:17 (00:0b:82:0e:e1:17)
.... 0. .... = IG bit: Individual address (unicast)
.... 0. .... = LG bit: Globally unique address (factory default)
Type: IP (0x0800)
Internet Protocol, Src: 192.168.104.246 (192.168.104.246), Dst: 192.168.104.204 (192.168.104.204)

```

Blue

```
Version: 4
Header length: 20 bytes
Differentiated Services Field: 0xc0 (DSCP 0x30: Class Selector 6; ECN: 0x00)
  1100 00.. = Differentiated Services Codepoint: Class Selector 6 (0x30)
  .... ..0. = ECN-Capable Transport (ECT): 0
  .... ...0 = ECN-CE: 0
Total Length: 635
Identification: 0x0002 (2)
Flags: 0x00
  0... = Reserved bit: Not set
  .0.. = Don't fragment: Not set
  ..0. = More fragments: Not set
Fragment offset: 0
Time to live: 255
Protocol: UDP (0x11)
Header checksum: 0x659c [correct]
  [Good: True]
  [Bad : False]
Source: 192.168.104.246 (192.168.104.246)
Destination: 192.168.104.204 (192.168.104.204)
User Datagram Protocol, Src Port: 5060 (5060), Dst Port: 5060 (5060)
Source port: 5060 (5060)
Destination port: 5060 (5060)
Length: 615
Checksum: 0x198c [correct]
Session Initiation Protocol
Request-Line: SUBSCRIBE sip:341@192.168.104.204 SIP/2.0
Method: SUBSCRIBE
  [Resent Packet: False]
Message Header
Via: SIP/2.0/UDP 192.168.104.246:5060;branch=z9hG4bK1970f9f3a5f432e4
  Transport: UDP
  Sent-by Address: 192.168.104.246
  Sent-by port: 5060
  Branch: z9hG4bK1970f9f3a5f432e4
From: "3COM 341" <sip:341@192.168.104.204>;tag=4095e80felfe02eb
  SIP Display info: "3COM 341"
  SIP from address: sip:341@192.168.104.204
  SIP tag: 4095e80felfe02eb
To: <sip:341@192.168.104.204>
  SIP to address: sip:341@192.168.104.204
Contact: <sip:341@192.168.104.246:5060;transport=udp>
  Contact Binding: <sip:341@192.168.104.246:5060;transport=udp>
  URI: <sip:341@192.168.104.246:5060;transport=udp>
  SIP contact address: sip:341@192.168.104.246:5060
Supported: eventlist, path
Call-ID: c3343d56b4d8503f@192.168.104.246
CSeq: 1001 SUBSCRIBE
  Sequence Number: 1001
  Method: SUBSCRIBE
User-Agent: Grandstream GXP2020 1.1.4.6
Max-Forwards: 70
Allow: INVITE,ACK,CANCEL,BYE,NOTIFY,REFER,OPTIONS,INFO,SUBSCRIBE,UPDATE,PRACK,MESSAGE
Event: message-summary
Expires: 3600
Accept: application/simple-message-summary
Content-Length: 0

Frame 9 (325 bytes on wire, 325 bytes captured)
Arrival Time: May 30, 2007 10:51:19.822030000
[Time delta from previous packet: 0.000754000 seconds]
[Time since reference or first frame: 0.124678000 seconds]
Frame Number: 9
Packet Length: 325 bytes
```

## Blue

```
Capture Length: 325 bytes
[Frame is marked: False]
[Protocols in frame: eth:ip:udp:sip]
[Coloring Rule Name: UDP]
[Coloring Rule String: udp]
Ethernet II, Src: Ibm_22:1a:1c (00:11:25:22:1a:1c), Dst: Grandstr_0e:e1:17 (00:0b:82:0e:e1:17)
  Destination: Grandstr_0e:e1:17 (00:0b:82:0e:e1:17)
    Address: Grandstr_0e:e1:17 (00:0b:82:0e:e1:17)
      .... 0... = IG bit: Individual address (unicast)
      .... 0... = LG bit: Globally unique address (factory default)
  Source: Ibm_22:1a:1c (00:11:25:22:1a:1c)
    Address: Ibm_22:1a:1c (00:11:25:22:1a:1c)
      .... 0... = IG bit: Individual address (unicast)
      .... 0... = LG bit: Globally unique address (factory default)
  Type: IP (0x0800)
Internet Protocol, Src: 192.168.104.204 (192.168.104.204), Dst: 192.168.104.246 (192.168.104.246)
  Version: 4
  Header length: 20 bytes
  Differentiated Services Field: 0xb8 (DSCP 0x2e: Expedited Forwarding; ECN: 0x00)
    1011 10.. = Differentiated Services Codepoint: Expedited Forwarding (0x2e)
      .... 0.. = ECN-Capable Transport (ECT): 0
      .... 0.. = ECN-CE: 0
  Total Length: 311
  Identification: 0x0000 (0)
  Flags: 0x04 (Don't Fragment)
    0... = Reserved bit: Not set
    1.. = Don't fragment: Set
    ..0 = More fragments: Not set
  Fragment offset: 0
  Time to live: 64
  Protocol: UDP (0x11)
  Header checksum: 0xe5ea [correct]
    [Good: True]
    [Bad : False]
  Source: 192.168.104.204 (192.168.104.204)
  Destination: 192.168.104.246 (192.168.104.246)
User Datagram Protocol, Src Port: 5060 (5060), Dst Port: 5060 (5060)
  Source port: 5060 (5060)
  Destination port: 5060 (5060)
  Length: 291
  Checksum: 0x716a [correct]
Session Initiation Protocol
  Status-Line: SIP/2.0 100 Trying
  Status-Code: 100
  [Resent Packet: False]
Message Header
  v: SIP/2.0/UDP 192.168.104.246:5060;branch=z9hG4bK1970f9f3a5f432e4
    Transport: UDP
    Sent-by Address: 192.168.104.246
    Sent-by port: 5060
    Branch: z9hG4bK1970f9f3a5f432e4
  f: "3COM 341"<sip:341@192.168.104.204>;tag=4095e80fe1fe02eb
    SIP Display info: "3COM 341"
    SIP from address: sip:341@192.168.104.204
    SIP tag: 4095e80fe1fe02eb
  t: <sip:341@192.168.104.204>
    SIP to address: sip:341@192.168.104.204
  i: c3343d56b4d8503f@192.168.104.246
  Cseq: 1001 SUBSCRIBE
    Sequence Number: 1001
    Method: SUBSCRIBE
  Date: Wed, 30 May 2007 13:50:07 GMT
  Content-Length: 0
```

Blue

```
Frame 10 (573 bytes on wire, 573 bytes captured)
  Arrival Time: May 30, 2007 10:51:19.822956000
  [Time delta from previous packet: 0.000926000 seconds]
  [Time since reference or first frame: 0.125604000 seconds]
  Frame Number: 10
  Packet Length: 573 bytes
  Capture Length: 573 bytes
  [Frame is marked: False]
  [Protocols in frame: eth:ip:udp:sip]
  [Coloring Rule Name: UDP]
  [Coloring Rule String: udp]
Ethernet II, Src: Ibm_22:1a:1c (00:11:25:22:1a:1c), Dst: Grandstr_0e:e1:17 (00:0b:82:0e:e1:17)
  Destination: Grandstr_0e:e1:17 (00:0b:82:0e:e1:17)
    Address: Grandstr_0e:e1:17 (00:0b:82:0e:e1:17)
      ....0... = IG bit: Individual address (unicast)
      ....0... = LG bit: Globally unique address (factory default)
  Source: Ibm_22:1a:1c (00:11:25:22:1a:1c)
    Address: Ibm_22:1a:1c (00:11:25:22:1a:1c)
      ....0... = IG bit: Individual address (unicast)
      ....0... = LG bit: Globally unique address (factory default)
  Type: IP (0x0800)
Internet Protocol, Src: 192.168.104.204 (192.168.104.204), Dst: 192.168.104.246 (192.168.104.246)
  Version: 4
  Header length: 20 bytes
  Differentiated Services Field: 0xb8 (DSCP 0x2e: Expedited Forwarding; ECN: 0x00)
    1011 10.. = Differentiated Services Codepoint: Expedited Forwarding (0x2e)
    ....0.. = ECN-Capable Transport (ECT): 0
    ....0.. = ECN-CE: 0
  Total Length: 559
  Identification: 0x0000 (0)
  Flags: 0x04 (Don't Fragment)
    0... = Reserved bit: Not set
    .1.. = Don't fragment: Set
    ..0. = More fragments: Not set
  Fragment offset: 0
  Time to live: 64
  Protocol: UDP (0x11)
  Header checksum: 0xe4f2 [correct]
    [Good: True]
    [Bad : False]
  Source: 192.168.104.204 (192.168.104.204)
  Destination: 192.168.104.246 (192.168.104.246)
User Datagram Protocol, Src Port: 5060 (5060), Dst Port: 5060 (5060)
  Source port: 5060 (5060)
  Destination port: 5060 (5060)
  Length: 539
  Checksum: 0x934a [correct]
Session Initiation Protocol
  Status-Line: SIP/2.0 200 OK
  Status-Code: 200
  [Resent Packet: False]
  Message Header
    v: SIP/2.0/UDP 192.168.104.246:5060;branch=z9hG4bK1970f9f3a5f432e4
      Transport: UDP
      Sent-by Address: 192.168.104.246
      Sent-by port: 5060
      Branch: z9hG4bK1970f9f3a5f432e4
    f: "3COM 341"<sip:341@192.168.104.204>;tag=4095e80fe1fe02eb
      SIP Display info: "3COM 341"
      SIP from address: sip:341@192.168.104.204
      SIP tag: 4095e80fe1fe02eb
    t: <sip:341@192.168.104.204>;tag=b3e36d3c
      SIP to address: sip:341@192.168.104.204
      SIP tag: b3e36d3c
```

Blue

```
i: c3343d56b4d8503f@192.168.104.246
Cseq: 1001 SUBSCRIBE
    Sequence Number: 1001
    Method: SUBSCRIBE
Date: Wed, 30 May 2007 13:50:07 GMT
m: <sip:3ComCallProcessor@192.168.104.204>
    Contact Binding: <sip:3ComCallProcessor@192.168.104.204>
        URI: <sip:3ComCallProcessor@192.168.104.204>
            SIP contact address: sip:3ComCallProcessor@192.168.104.204
Allow: INVITE,ACK,BYE,CANCEL,REFER,SUBSCRIBE,NOTIFY,UPDATE,OPTIONS,MESSAGE,FEATURE
k: eventlist, path
Expires: 3600
Event: message-summary
User-Agent: 3Com VCX 7210 IP CallProcessor/v7.1.42
Content-Length: 0
```

**Frame 11 (533 bytes on wire, 533 bytes captured)**

```
Arrival Time: May 30, 2007 10:51:19.843818000
[Time delta from previous packet: 0.020862000 seconds]
[Time since reference or first frame: 0.146466000 seconds]
Frame Number: 11
Packet Length: 533 bytes
Capture Length: 533 bytes
[Frame is marked: False]
[Protocols in frame: eth:ip:udp:sip]
[Coloring Rule Name: UDP]
[Coloring Rule String: udp]
Ethernet II, Src: Grandstr_0e:e1:17 (00:0b:82:0e:e1:17), Dst: Ibm_22:1a:1c (00:11:25:22:1a:1c)
  Destination: Ibm_22:1a:1c (00:11:25:22:1a:1c)
    Address: Ibm_22:1a:1c (00:11:25:22:1a:1c)
      .... 0... = IG bit: Individual address (unicast)
      .... 0... = LG bit: Globally unique address (factory default)
  Source: Grandstr_0e:e1:17 (00:0b:82:0e:e1:17)
    Address: Grandstr_0e:e1:17 (00:0b:82:0e:e1:17)
      .... 0... = IG bit: Individual address (unicast)
      .... 0... = LG bit: Globally unique address (factory default)
Type: IP (0x0800)
Internet Protocol, Src: 192.168.104.246 (192.168.104.246), Dst: 192.168.104.204 (192.168.104.204)
Version: 4
Header length: 20 bytes
Differentiated Services Field: 0xc0 (DSCP 0x30: Class Selector 6; ECN: 0x00)
  1100 00.. = Differentiated Services Codepoint: Class Selector 6 (0x30)
  .... ..0. = ECN-Capable Transport (ECT): 0
  .... ...0 = ECN-CE: 0
Total Length: 519
Identification: 0x0003 (3)
Flags: 0x00
  0... = Reserved bit: Not set
  .0.. = Don't fragment: Not set
  ..0. = More fragments: Not set
Fragment offset: 0
Time to live: 255
Protocol: UDP (0x11)
Header checksum: 0x660f [correct]
  [Good: True]
  [Bad : False]
Source: 192.168.104.246 (192.168.104.246)
Destination: 192.168.104.204 (192.168.104.204)
User Datagram Protocol, Src Port: 5060 (5060), Dst Port: 5060 (5060)
Source port: 5060 (5060)
Destination port: 5060 (5060)
Length: 499
Checksum: 0x12f9 [correct]
Session Initiation Protocol
```

## Blue

```
Status-Line: SIP/2.0 200 OK
Status-Code: 200
[Resent Packet: False]
Message Header
Via: SIP/2.0/UDP 192.168.104.204;branch=z9hG4bK8011f557-220d-dc11-887a-b53b2d69a9ca
Transport: UDP
Sent-by Address: 192.168.104.204
Branch: z9hG4bK8011f557-220d-dc11-887a-b53b2d69a9ca
From: <sip:192.168.104.204>;tag=b3e667d4
SIP from address: sip:192.168.104.204
SIP tag: b3e667d4
To: <sip:341@192.168.104.204>;tag=06e5defe37feec1f
SIP to address: sip:341@192.168.104.204
SIP tag: 06e5defe37feec1f
Call-ID: 80b690e7-750c-dc11-a771-d53e931519da
CSeq: 51 NOTIFY
Sequence Number: 51
Method: NOTIFY
User-Agent: Grandstream GXP2020 1.1.4.6
Contact: <sip:341@192.168.104.246:5060;transport=udp>
Contact Binding: <sip:341@192.168.104.246:5060;transport=udp>
URI: <sip:341@192.168.104.246:5060;transport=udp>
SIP contact address: sip:341@192.168.104.246:5060
Allow: INVITE,ACK,CANCEL,BYE,NOTIFY,REFER,OPTIONS,INFO,SUBSCRIBE,UPDATE,PRACK,MESSAGE
Supported: replaces, timer
Content-Length: 0
```