GRANDSTREAM

CONNECTING THE WORLD

voip SUPPLY

How to secure your phone calls with UCM6200 & GXP phones

AGENDA

- Founded in 2002
- 30+ employees
- Over 125,000 customers worldwide

Office: Amherst, New York
Contact Us: 1-800-398-VoIP

AGENDA

Since 2002 VoIP Supply has delivered unparalleled service and expertise to over 125,000 customers worldwide.

NEW VoIP Rental Program
our Device as a Service (DaaS) rental program gives your customers the widest variety of VoIP Products for a low monthly payment!

Hardware
featuring over 60 manufacturers that offer over 16,000 products

Experts on Your Side
the most qualified professionals become your extended team!

CloudSpan MarketPlace
a single place to shop various VoIP service providers - finding the perfect match for your client's diverse business needs

Fulfillment
provisioning and professional services from multiple warehouse locations in North America; providing real-time access to manage your projects from order through delivery

Refresh & Reclaim
offering certified reconditioned devices at a fraction of the cost plus offering an outlet for selling off used and excess VoIP equipment

AGENDA

The Simplest way to sell cloud services & make $

1. Sign VoIP Supply's CloudSpan Reseller Agreement
2. Introductions & training w/CloudSpan vendors
3. Leverage resources for marketing & quoting
4. Cash your commission checks!

Benefits of CloudSpan

- No quotas/commitment levels
- Best in class commissions and spiffs/incentives
- Sell hardware + services
- Programs for single person through SMB to enterprise
- Commissions grow as your business grows,
  higher than if you go direct

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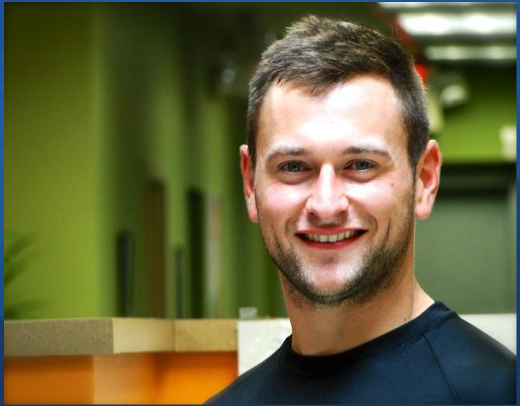
Become a VoIP Supply Partner

Call or Email Jon Garbin at *(716) 531- 4271 jgarbin@voipsupply.com*

- Complete partner program agreements & paperwork
- Create an Onboarding Game Plan with your Account Representative (Training & Equipping your Team)
- Marketing & Sales Collateral - https://www.voipsupply.com/partner-portal-home-page/
- Opportunity Support – Consulting & Solution Design, Proposal/Pricing



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1. INTRODUCTION
2. VOIP FUNDAMENTALS
3. SIGNALING VS MEDIA
4. ENCRYPTING YOUR SIGNALING WITH TLS
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VoIP Phone Product Category

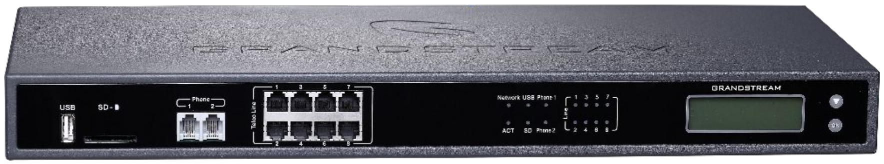
UCM6200 series

IP PBX Appliance

- 2/4/8 FXO trunk port models, 2 FXS ports with lifeline capability.
- Up to 100 concurrent calls and up to 32 conference attendees
- Up to 800 SIP endpoints and up to 50 SIP trunk accounts
- Dual Gigabit network ports, integrated PoE, USB and SD ports
- Zero Configuration endpoint provisioning and no licensing fees

UCM6200 series

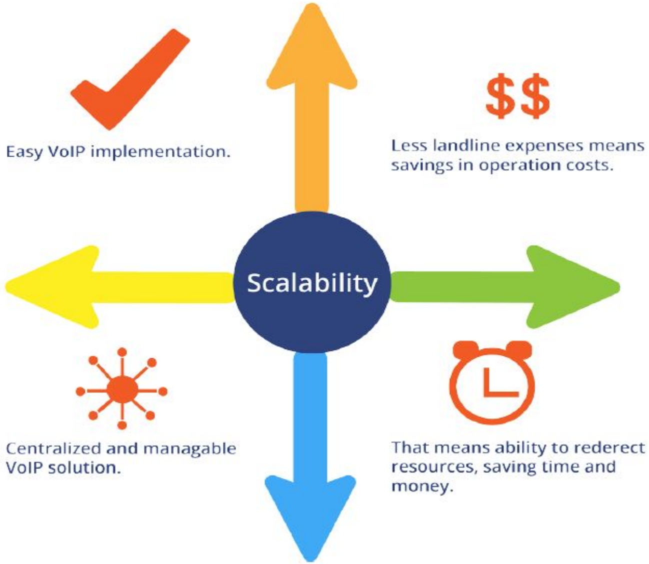
Enhanced Features



- Up to 100 concurrent calls and up to 32 conference attendees
    - 50 concurrent calls – UCM6202
    - 75 concurrent calls – UCM6202
    - 100 concurrent calls – UCM6208
- Up to 800 SIP endpoints
- Dual-core 1GHz processor, 1GB RAM for enhanced capacity and speed
- Three sizes – UCM6202, 6204 and 6208

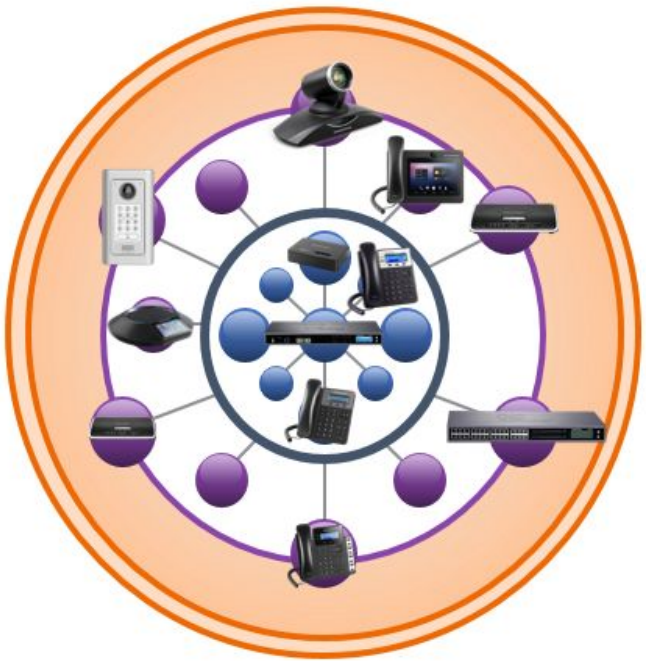
VoIP Fundamentals

VoIP offers businesses many attractive benefits, one of the most alluring is a reduction in the overall cost of communicating. The business world is increasingly global, and businesses need to keep up and stay accessible



Grandstream Solutions: UC Components

- Communications: Voice, data, and video
- Messaging: Voice, email, video, and IM
- Conferencing: Online, audio, and video
- Application integration: Microsoft Office and CRM
- Presence: IP phone, desktop clients, and call connectors
- Common user experience: Desktop, phone, and mobility

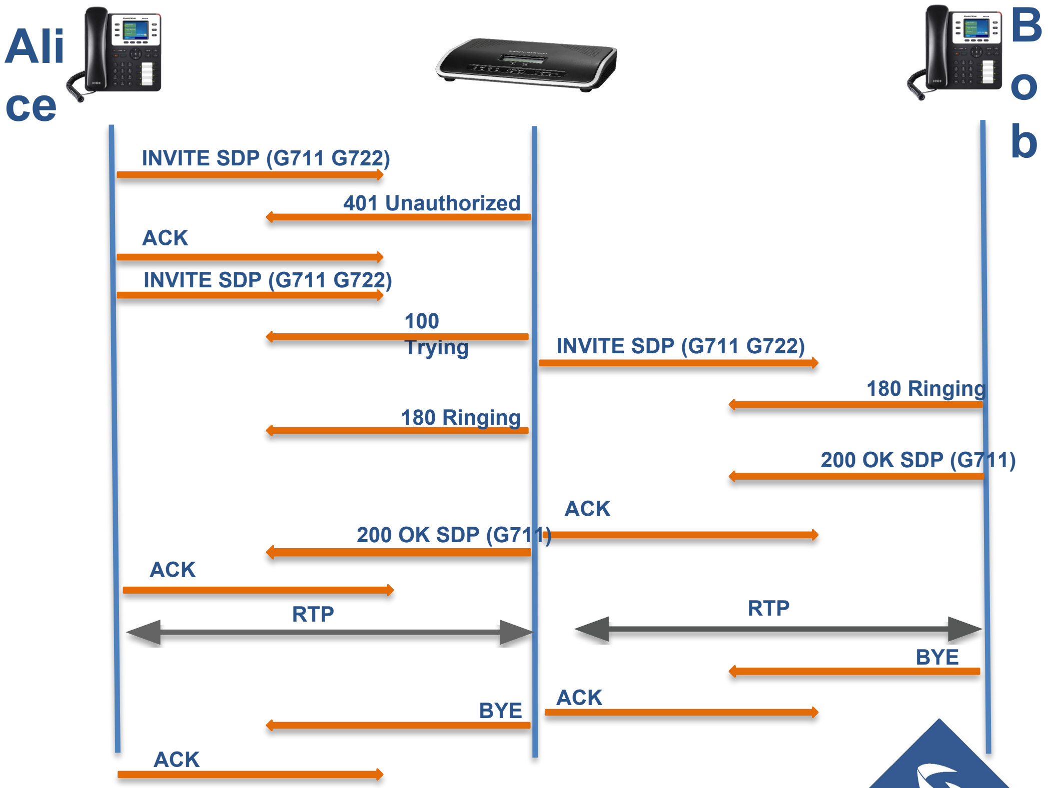


Signaling

Session Initiation Protocol

- The **Session Initiation Protocol** (**SIP**) is a signaling protocol used in VoIP communication to set up, control, and terminate call sessions
- SIP works together with **SDP** to specify and carry the session media type and negotiate the media parameters
- SIP is used with other protocols like **RTP** and **RTSP** to provide a complete multimedia architecture

Call Flow

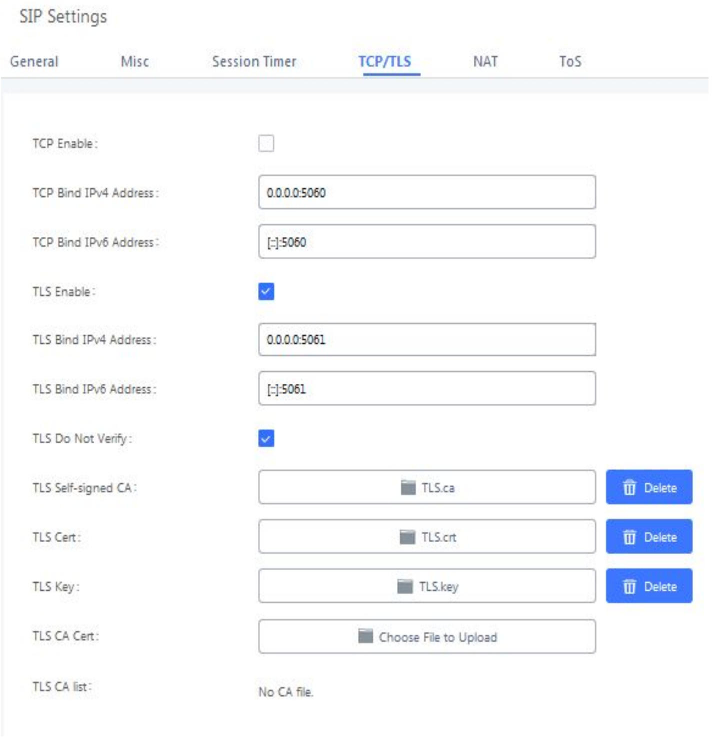



```
Request-Line: INVITE sip:1006@192.168.22.119 SIP/2.0
Message Header
  Via: SIP/2.0/UDP 192.168.22.58:5060;branch=z9hG4bK712675994;rport
  From: "Steve Common" <sip:1005@192.168.22.119>;tag=1423635195
  To: <sip:1006@192.168.22.119>
  Call-ID: 25873442-5060-7@BJC.BGI.CC.FI
  CSeq: 20 INVITE
  Contact: "Steve Common" <sip:1005@192.168.22.58:5060>
  X-Grandstream-PBX: true
  Max-Forwards: 70
  User-Agent: Grandstream GXP2170 1.0.9.69
  Privacy: none
  P-Preferred-Identity: "Steve Common" <sip:1005@192.168.22.119>
  P-Emergency-Info: IEEE-EUI-48;eui-48-addr=00-0B-82-6F-8E-EE
  Supported: replaces, path, timer
  Allow: INVITE, ACK, OPTIONS, CANCEL, BYE, SUBSCRIBE, NOTIFY, INFO, REFER, UPDATE, MESSAGE
  Content-Type: application/sdp
  Accept: application/sdp, application/dtmf-relay
  Content-Length:   435
Message Body
  Session Description Protocol
    Session Description Protocol Version (v): 0
    Owner/Creator, Session Id (o): 1005 8000 8000 IN IP4 192.168.22.58
    Session Name (s): SIP Call
    Connection Information (c): IN IP4 192.168.22.58
    Time Description, active time (t): 0 0
    Media Description, name and address (m): audio 5008 RTP/AVP 0 8 4 18 9 97 2 123 101
    Media Attribute (a): sendrecv
    Media Attribute (a): rtpmap:0 PCMU/8000
    Media Attribute (a): ptime:20
    Media Attribute (a): rtpmap:8 PCMA/8000
    Media Attribute (a): rtpmap:4 G723/8000
    Media Attribute (a): rtpmap:18 G729/8000
    Media Attribute (a): fmtp:18 annexb=no
    Media Attribute (a): rtpmap:9 G722/8000
    Media Attribute (a): rtpmap:97 iLBC/8000
    Media Attribute (a): fmtp:97 mode=30
    Media Attribute (a): rtpmap:2 G726-32/8000
    Media Attribute (a): rtpmap:123 opus/48000/2
    Media Attribute (a): rtpmap:101 telephone-event/8000
```

Supported media

Session Initiation Protocol Secure (SIPS)

- SIP messages are passed in clear text which makes the protocol vulnerable to certain attacks
- For secure transmissions of SIP messages over an unprotected network, the protocol may be encrypted with TLS
- Server authentication via TLS Certificate

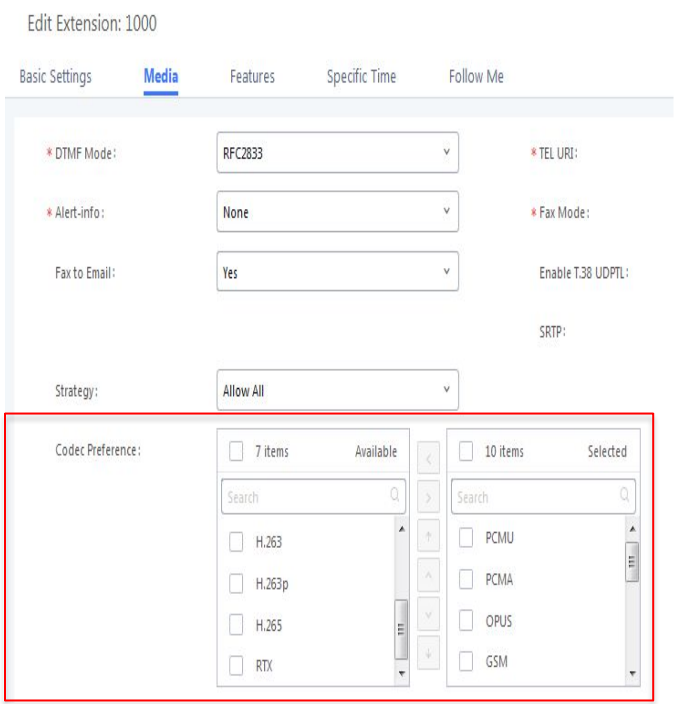


Media

Real-Time Transport Protocol

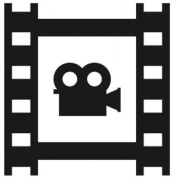
- Real-time Transport protocol (RTP) provides end-to-end delivery for audio traffic over IP network
- RTP runs on top of UDP
- RTP does not have a mechanism to prevent out-of-order packet delivery which might cause jitter in congested networks
- RTP is used in conjunction with RTCP to provide statistics and QoS

Voice Codec



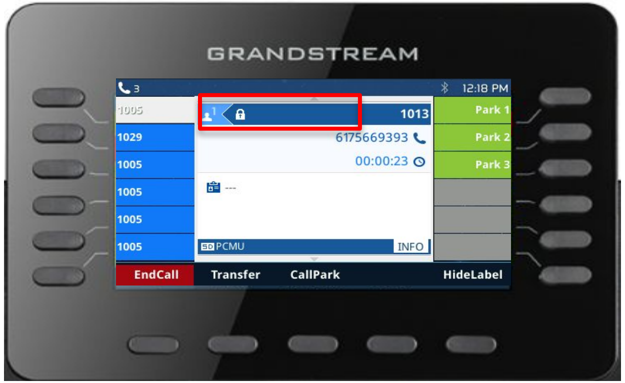
Voice codecs are used to encode audio signal into digital data, and vice versa

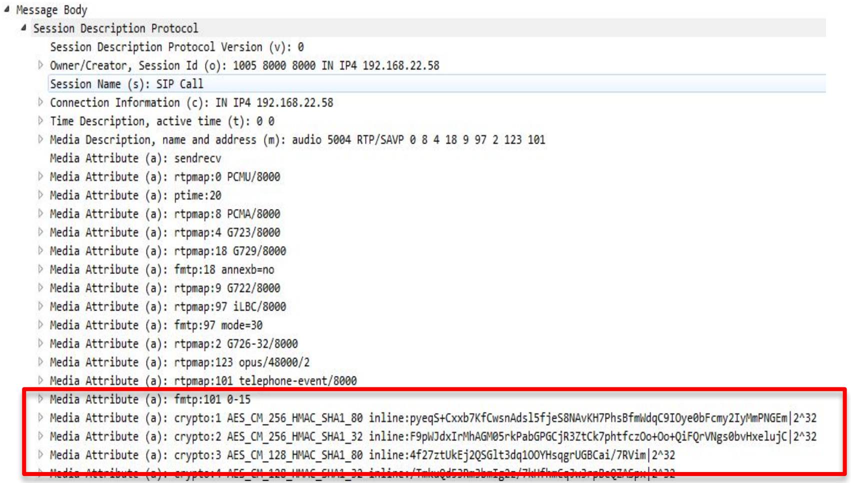
- **G.711** uses 64 kbps for one-way audio and less resource intensive
- **G.729** uses 8 kbps which makes it excellent for bandwidth optimization but more resource intensive
- **G.722** allows for bitrates of 64, 56, and 48 kbps.
- **iLBC** operates at 15 kbps
- **H.264** is an industry standard for video compressi



SRTP

- Secure Real-time Transport protocol (SRTP) provides encryption, authentication and anti-replay
- SRTP uses AES 128/256 bit to encrypt audio traffic
- The crypto-key used for SRTP is exchanged in the SDP message
- Grandstream devices support two modes of SRTP:
  - Enabled but Not Forced
  - Enabled and Forced





GRANDSTREAM

CONNECTING THE WORLD

Live Demo

How to Secure UCM6000 & GXP2100

Questions?

Please submit your questions using the Q/A feature.

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