GRANDSTREAM
CONNECTING THE WORLD

DP755

The DP755 is a powerful VoIP phone base station that pairs with up to 10 of Grandstream's DP series. It supports DP730/DP725/DP735/ DP722/ DP720. This VoIP phone supports up to 20 SIP accounts while also offering 3-way voice conferencing, full HD audio and integrated PoE. A shared SIP account on all handsets will add seamless unified features that gives users the ability to answer all calls regardless of location in real-time. The DP755 supports a variety of auto-provisioning methods and TLS/SRTP/HTTPS encryption security. When paired with Grandstream's, the DP755 offers a powerful solution for any business or residential user.

20 LINES

Up to 20 SIP accounts per system; up to 20 lines

DECT authentication & encryption technology to protect calls & account

3 WAY

3-way audio conferencing for easy conference calls

Supports Push-to-Talk

Automated provisioning options include TR-069 and XML config files

zero CONFIG

Use with Grandstream's UCM series of IP PBXs for Zero Config provisioning

HD

HD audio to maximize audio quality and clarity; full duplex speakerphone

Supports advanced telephony features, including call transfer, call forward, call-waiting, do not disturb, message waiting indication, multi-language promtps, flexible dial plan and more

Peripherals	3 LED indicators: Power, Network, One 10/100 Mbps auto-sensing Ethernet port with integrated PoE
Protocol/Standards	SIP RFC3261, TCP/IP/UDP, RTP/RTCP, HTTP/HTTPS, ARP/RARP, ICMP, DNS (A record, SRV, NAPTR), DHCP, PPPoE, SSH, TFTP, NTP, STUN, SIMPLE, LLDP-MED, LDAP, TR-069, 802.1x, TLS, SRTP, IPv6
Voice Codecs	G.711μ/a-law, G.723.1, G.729A/B, G.726-32, iLBC, G.722, OPUS, in-band and out-of-band DTMF (in audio, RFC2833, SIP INFO), VAD, CNG, PLC, AJB
Telephony Features	Hold, transfer, forward, 3-way conference, downloadable phonebook (XML, LDAP, up to 3000 entries), call waiting, call log (up to 1500 records), auto answer, flexible dial plan, server redundancy and fail-over
QoS	Layer 2 QoS (802.1Q, 802.1P) and Layer 3 QoS (ToS, DiffServ, MPLS)
Security	User and administrator level access control, MD5 and MD5-sess based authentication, 256-bit AES encrypted configuration file, TLS, SRTP, HTTPS, 802.1x media access control, DECT authentication & encryption
Multi-language	English, Czech, German, Spanish, French, Hebrew, Italian, Dutch, Polish, Portuguese, Russian, Turkish, Arabic, Chinese Simple, Chinese Tradition, Japanese, Korean, Slovakian, Serbian
Upgrade/ Provisioning	Firmware upgrade via TFTP/HTTP/HTTPS and local uploading, mass provisioning using TR-069 or AES encrypted XML configuration file
Multiple SIP Accounts	Up to twenty (20) distinct SIP accounts per system **Each handset** may map to up to 20 SIP account(s) **Each SIP account** may map to any handset(s)
Ring Group	Flexible options when multiple handsets share the same SIP account: **Parallel Mode:** all phones ring concurrently and after one phone answers, the remaining available phones can make new calls **HS Mode:** multiple handsets can call out on the same account, but all incoming calls are directed to a single handset
Power & Green Energy Efficiency	Universal Power Supply: Input AC 100-240V 50/60Hz; Output 5VDC 1A; Micro-USB connection; PoE: IEEE802.3af Class 1, 0.44W–3.84W
Package Content	Base Unit, Universal Power Supply; Ethernet cable; Quick Start Guide, GPL statement
Dimensions (H x W x D)	140.31 x 64.98 x 105mm
Weight	Base unit: 140g; Universal power supply: 50g; Package: 370g
Temperature and Humidity	Operation: -10 to 45ºC (14 to 113ºF); Storage: -20º to 60ºC (-4 to 140ºF) Humidity: 10% to 90% non-condensing
Compliance	FCC, CE, RCM, IC