GRANDSTREAM
CONNECTING THE WORLD

DP715/710

DECT Cordless IP Phones for Mobility

These compact and durable DECT IP phones allow users mobility throughout their home or office while maintaining the benefits of VoIP calling. Users can register up to 5 DECT handsets to a single base station, which supports 4 concurrent calls. The DECT base station is inside of the DP715. When multiple handsets share the same SIP account, users can take advantage of the ring group/hunt group feature to make sure that all calls are always answered. By supporting a range of 300 meters outdoors and 50 meters indoors, the DP715/710 offer a great cordless IP phone for office buildings, retail stores, warehouse and more.

5 LINES
Up to 5 SIP accounts per system, independent SIP account per handset

TLS and SRTP security encryption technology to protect calls and accounts

3 WAY
3-way audio conferencing for easy conference calls

SIP
DP715/710 are fully compliant with the SIP/ DECT standard

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

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

Air Interface	**Telephony standards:** DECT / GAP (frequency range: 1880 ~ 1900 MHz (Europe), 1920 ~ 1930 MHz (US), 1910 ~ 1920 MHz (Brazil) **Number of Channels:** 120 (Europe) or 60 (US and Brazil) duplex channels **Emission power:** 10mW (average power per channel) **Range:** up to 300 meters outdoors and 50 meters indoors
Network Interfaces	One 10/100Mbps auto-sensing ethernet port (RJ45, Base station only)
LED Indicators	Base station: Power, Network, Register, Call
Handset Display	1.7 inch 102x80 LCD with blue backlight
Voice over Packet Capabilities	Base station: Dynamic jitte buffer Handset: Speakerphone with acoustic echo cancellation
Voice Compression	G.711 with Annex I (PLC) and Annex II (VAD/CNG), G.723.1, G.726-32 AAL2, G.729A/B, iLBC
Telephony Features	Caller ID display or block, call waiting, flash, blind or attended transfer, forward, hold, do not disturb, 3-way conference
QoS	Layer 2 (802.1Q VLAN/802.1p), Layer 3 (ToS, Diffserv, MPLS)
IP Transport	RTP/RTCP
DTMF Method	In-Audio, RFC2833 and SIP INFO
IP Signaling	SIP (RFC 3261)
Multiple SIP accounts per base station	Up to 5 distinct SIP accounts per system, independent SIP account per handset, multiple handsets per SIP account
Hunting Group	Linear mode, Parallel mode, Shared Line mode
Provisioning	HTTP, HTTPS, TFTP, TR-069, secure and automated provisioning
Security	TLS, SRTP, HTTPS
Management	Web interface or via secure (AES encryoted) central configuration file for mass deployment, supports device conifguration via built-in IVR, web browser or central configuration file through TFTP, HTTP or HTTPS
Phonebook (per handset)	200 records each with up to 24 digits and 16 characters 10 dialed call records, 20 recieved call records
Multi-Language Display (handset LCD menu)	English, German, Italian, French, Spanish, Portuguese, Dutch, Czech, Danish, Greek, Norwegian, Russian, Polish, Swedish, Turkish
Polyphonic Ringtones	18 ringtones available
Universal Power Supply	Input: 100-240VAC, 50-60Hz, 0.15A; Output: 6VDC, 500mA (for base station); 7VDC, 420mA (for charger unit)
Battery Life	10 hour talk time, 80 hour stand-by time, 16 hourcharging time
Dimensions (H x W x D)	Base station: 75 x 105 x 85 mm; Handset: 160 x 46 x 22 mm; Charging unti: 53 x 75 x 60 mm
Weight	Base station: 95g Handset (with batteries): 104g Charging unit: 102g
Environmental	Operational: 32° ~ 104°F or 0° ~ 40°C; Storage: -4° ~ 140°F or -20°~ 60°C; Humidity: Maximum 85% non condensing
Compliance	FCC Part 15B/15D; CE: ETSI EN 301 489-1 V1.8.1, ETSI EN 301 489-6 V1.3.1, ETSI EN 301 406-1 V2.1.1, ETSI EN 301 406 V1.5.1, EN60950-1, RoHs, UL (power supply)