



## PANASONIC KX-UT SERIES SIP TERMINALS FEATURE SHEET

The Panasonic UT range of SIP telephony terminals enhance personal communications through excellent HD quality audio on every phone, combined with easy access to powerful supportive features and applications. The range - from standard phones, office key-sets, executive terminals and touch-screen Smart Desk application phones - addresses all requirements. Panasonic's reputation for design, quality, reliability and care for the environment, ensures an exceptional user experience wherever the terminals are deployed - as part of a "cloud based" service or with an IP PBX - in a business environment or in the home.



### KEY FEATURES

#### High Definition "HD" Audio

The KX-UT series of SIP terminals offers 'best in class' audio quality, meaning fewer repeated conversations and misheard calls. Offering Wideband High Definition Audio as standard across the range, the UT series offer G.722, G.711, G.726 and G.729a Codecs. Coupled with Enhanced Echo Cancellation and an Expanded Acoustic Chamber, the UT series of SIP terminals offers a superior audio experience to users over handset, speakerphone and optional headsets. The entire range features wideband compliant, hearing aid compatible handsets, and built-in, high quality speaker and microphone.

#### LCD Displays

Large, clear LCD displays with intuitive User Interface offer fast access to phonebooks and features.

#### Electric Hook Switch

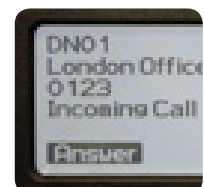
A built in Electronic Hook Switch (Plantronics compliant) port allows the KX-UT133 and KX-UT136 SIP Terminals to have access to the Plantronics range of DECT enabled headsets. This offers a range of portability and comfort as frequent users are able to move around freely, without being tied down by handsets.

#### ECO Friendly

Low power consumption, combined with an advanced ECO standby mode means lower energy costs. Consumption can be less than 1 watt in ECO mode! (UT 1xx series).

#### Plug and Play Configuration

Extensive provisioning options, including automatic configuration via a provisioning server, means a smaller administration overhead, saving time and money. The range is certified compatible with Digium Asterisk and Broadsoft Broadworks, ensuring excellent compatibility with leading soft switch suppliers.





	KX-UT113	KX-UT123	KX-UT133	KX-UT136	KX-UT248*	KX-UT670*
<b>DISPLAY</b>						
LCD Display	Monochrome Graphical	Monochrome Graphical	Monochrome Graphical	Monochrome Graphical	Greyscale Graphical	Colour 262k
LCD Size	242 x 55 pixels – 3 lines			242 x 109 pix. – 6 lines	4.4 inch	7 inch touch
LCD Contrast	6 levels					No
LCD Backlight	None	On/Auto/Off				15/Auto
HD Video	No					720p
<b>INSTALL OPTIONS</b>						
Desk mount tilt	No		Yes – 2 positions			
Wall mount	KX-A432 (optional)		KX-A433 (optional)			KX-A434 (optional)
Power adaptor	Option - KX-A239				Option - KX-A422	
<b>NETWORKING</b>						
Ethernet Ports	1 – 10/100	2 – 10/100			2 – 10/100/1000	
Power over Ethernet (PoE)	Yes					
Bluetooth	No				Yes	Optional
<b>AUDIO FEATURES</b>						
Handset, Speaker, Headset Volume	8 levels (includes echo cancellation and distortion prevention)					15 levels
Ringtones	32					
Ringer Volume	6 levels + Off				7 levels + Off	
Headset Port	2.5 mm					
Electronic Hook Switch Control Port	None		3.5 mm Plantronics compatible			
Audio Codec	G.711, G.722, G.726, G.729a					G.711, G.722, G.729a
<b>KEYS</b>						
Soft Keys	Yes - 4					Yes - screen
Programmable Keys	0	24			24 (3 pages of 8 keys)	24 (4 pages of 6 keys)
0-9, *, # keys	Yes					Yes - screen keys
Navigator & Cancel Key	Yes					On screen + menu keys
<b>SOFTWARE FEATURES</b>						
Phone Book (Entries)	100 – each with 5 numbers	500 - each with 5 numbers				Depends on memory *2
Call Log Entries	None	30 incoming calls + 30 outgoing calls				100 in + 100 out
Conferencing	3 parties (within terminal – multi-party dependent on server)					
XML	XML application support from Q1 2012					
Music on Hold	Supplied by Host Service (PBX / SIP Server)					
<b>IP FEATURES</b>						
SIP Accounts	2	4			6	
SIP Compatibility	RFC 3261 Standard SIP Server, Asterisk, Broadsoft, Panasonic SIP PBX					
IP Version	IPv4					
DHCP Client	Yes					
DNS	Yes					
HTTP	Yes					
HTTPS	Yes					
SNTP Client	Yes					
VLAN (802.1q)	Yes					
QoS (DiffServ)	Yes					
802.1x	No				Yes	
<b>CONFIGURATION</b>						
Plug & Play Configuration	Server based configuration, TR-069, Panasonic Redirect Server					
Manual Configuration	Internal web Configurator, Local (LCD based) network configuration					

\*Preliminary Specifications  
\*2 up to 500 connected to NS1000