NetVanta Unified Communications Technical Note

NetVanta UC Server Interoperable SIP Device Features and Comparisons

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1 ADTRAN Connect™ Interoperability Program

ADTRAN has established the ADTRAN Connect™ Interoperability Program to allow participants in the program (Partners) to certify that their products and/or services are interoperable with ADTRAN's products: NetVanta Unified Communications Server™ and CallAttendant Office™. The program also allows for interoperability certification between third-party products that hold an ADTRAN Interoperability Mark, when ADTRAN recommends combinations of third-party products, such as headsets for use with telephones, or telecommunications services for use with gateways. The program covers the following product categories:

- Telephones (wired, wireless, soft, cellular, communicators)
- Gateways (analog PSTN, digital PSTN, ALGs, SBCs, etc.)
- Server Hardware Platforms
- Communications Cards
- Audio Devices (headsets [wired, wireless], USB telephones)
- Private Branch Exchanges (SIP, Legacy)
- Application Servers (voice mail, e-mail, IVR, call center, conference, etc.)
- Reporting Applications/Services (call accounting, billing, etc.)
- Application Software
- Telecommunications Services (ITSP, PSTN, etc.)

Organizations that enter into the ADTRAN Connect™ Interoperability Program Agreement become a recognized ADTRAN Connect™ Program Partner and may refer to themselves as an ADTRAN Connect™ Program Partner in their promotional literature, press releases and on their Web site.

Partners with at least one Partner Product that has been granted the ADTRAN Connect PLUS™ Interoperability Mark may refer to themselves as an ADTRAN Connect PLUS™ Program Partner in their promotional literature, press releases and on their web site.

The Program acknowledges the interoperability of Partner Products with ADTRAN Products by awarding Interoperability Marks to the Partner Products. Two different Interoperability Marks can be awarded to Partner Products depending on their level of integration with ADTRAN Products.

Capability	Connect Plus	Connect	Interoperable
Business relationship established	Yes	Yes	No
ADTRAN 2nd line support provided	Yes	Yes	No
Escalation from ADTRAN to partner	Yes	Yes	No
Configuration templates included	Yes	No	Maybe
Automatic detection	Yes	No	Maybe
Provisioning capable	Yes	No	Maybe
Configuration change notification	Yes	No	Maybe
Automatic firmware upgrade	Yes	No	No

1.1.1 ADTRAN Connect™ Mark

The ADTRAN Connect™ Interoperability Mark is awarded to products that interoperate with either ADTRAN Products or third-party products that hold an Interoperability Mark under the Program when: either no configuration of the Partner Product is required, or configuration of the Partner Product is achieved using Partner supplied configuration tools and methods according to ADTRAN provided Integration Notes. Products that are awarded this mark typically do not require ADTRAN to modify its products to achieve interoperability. This mark is intended to cover all the program's product categories.

Partners may describe Partner Products that have been awarded this mark as "ADTRAN Connect™ Compatible" and may employ the ADTRAN Connect™ logo in conjunction with the product, in their marketing programs, advertising, and other communications. Partners may also affix the logo to the Partner Product, its labeling and/or packaging materials, and may indicate that the Partner Product has been successfully tested under the ADTRAN Connect™ Interoperability Program.

1.1.2 ADTRAN Connect PLUS™ Mark

The ADTRAN Connect PLUS™ Interoperability Mark is awarded only to products that interoperate directly with ADTRAN products and only when the ADTRAN product makes special provisions to detect/locate and substantially configure a partner product. Products that are awarded this mark require ADTRAN to modify its products to achieve this level of interoperability. Automatic configuration is limited to a Standard Feature Set intended to cover the operational requirements of most installations. Additional features must be configured according to the partner's product configuration instructions.

Partners may describe Partner Products that have been awarded this mark as "ADTRAN Connect PLUS™ Compatible" and may employ the ADTRAN Connect PLUS™ logo in conjunction with the product, in their marketing programs, advertising, and other communications. Partners may also affix the logo to the Partner Product, its labeling and/or packaging materials, and may indicate that the Partner Product has been successfully tested under the ADTRAN Connect PLUS™ Interoperability Program.

This mark is intended to cover only the following product categories:

- Telephones (wired, wireless, soft)
- Gateways (analog PSTN, digital PSTN, Session Border Controllers, Application Layer Gateways)
- Communications Cards
- Telecommunications Services

1.1.3 ADTRAN Interoperable

The Program acknowledges the interoperability of third party products with ADTRAN Products by awarding the ADTRAN Interoperable™ Mark to the third party product. The ADTRAN Interoperable™ Mark may be awarded to products that interoperate with either ADTRAN Products or other third-party products that hold an ADTRAN Connect™ Interoperability Mark when: either no configuration of the third party product is required, or configuration of the third party product is achieved using the third party vendor supplied configuration tools and methods according to ADTRAN provided Integration Notes. Products that are awarded this mark do not usually require ADTRAN to modify its products to achieve interoperability.

ADTRAN interoperable products do not imply a business relationship with the vendor.

2 Desktop SIP Telephones

2.1 ADTRAN

Capability	*IP706*	*IP712*
Integration level	Connect+	Connect+
Supported firmware version	**1.3.11**	**1.3.11**
Automatic detection	√[1]	√[1]
Automatic provisioning	√	√
Automatic firmware upgrade	√	√
Max # of SIP registrations	6	12
LCD screen size*	6x35	9x35
Available codecs**	1,2,3	1,2,3
Programmable hard keys	6	12
Programmable soft keys		
Offhook dialing		
Offhook dialing & commit		
One button message retrieval	√	√
Per-identity message retrieval		
Per-identity MWI		
Call lists – missed calls	√	√
Call lists – received calls	√	√
Call lists – placed calls	√	√
Handset operation	√	√
Headset operation	√	√
Speakerphone – full duplex	√	√
Call recording		
Speed dial	√	√
PC Ethernet port	√	√
Mute	√	√
Directory-enabled dialing		
DND	√	√
PoE (Power over Ethernet)	√	√
SRTP		
TLS		
Backup proxy/registrar		
Paging capable	√[2]	√[2]

* ^ (lines by characters); **x** (pixels)
** Codecs: 1 (G.711µ), 2 (G.711A), 3 (G.729A)
[1] Phones in a different subnet than the UC server can also be automatically detected
[2] Supports unicast paging

2.2 Polycom SoundPoint IP (HD Voice™ capable models)

Capability	*450*	*550*	*560*	*650*	*670*
Integration level	Connect+	Connect+	Connect+	Connect+	Connect+
Supported firmware version	3.1.2.0392	3.1.2.0392	3.1.2.0392	3.1.2.0392	3.1.2.0392
Automatic detection	√[3]	√[3]	√[3]	√[3]	√[3]
Automatic provisioning	√	√	√	√	√
Automatic firmware upgrade	√	√	√	√	√
Max # of SIP registrations	3	4	4	6[1]	6[1]
LCD screen size*	256x116	320x160	320x160	320x160	320x160
Available codecs**	1,2,3,4	1,2,3,4	1,2,3,4	1,2,3,4	1,2,3,4
Programmable hard keys					
Programmable soft keys	2	4	4	6	6
Offhook dialing	√	√	√	√	√
Offhook dialing & commit	√	√	√	√	√
One button message retrieval	√	√	√	√	√
Per-identity message retrieval	√	√	√	√	√
Per-identity MWI	√	√	√	√	√
Call lists – missed calls	√	√	√	√	√
Call lists – received calls	√	√	√	√	√
Call lists – placed calls	√	√	√	√	√
Handset operation	√	√	√	√	√
Headset operation	√	√	√	√	√
Speakerphone – full duplex	√	√	√	√	√
Call recording	√	√	√	√	√
Speed dial	√	√	√	√	√
PC Ethernet port	√	√	√	√	√
Mute	√	√	√	√	√
Directory-enabled dialing	√	√	√	√	√
DND	√	√	√	√	√
PoE (Power over Ethernet)	√	√	√	√	√
SRTP					
TLS	√[2]	√[2]	√[2]	√[2]	√[2]
Backup proxy/registrar	√	√	√	√	√
Paging capable	√[4]	√[4]	√[4]	√[4]	√[4]

* ^ (lines by characters); **x** (pixels)
** Codecs: 1 (G.711µ), 2 (G.711A), 3 (G.729A), 4 (G.722)
[1] Can register up to 12 identities with expansion module
[2] Not currently supported by NetVanta Unified Communications Server
[3] Phones in a different subnet than the UC server can also be automatically detected
[4] Supports unicast paging

2.3 Polycom SoundPoint IP (all others)

Capability	*320/330*	*430*	*501*	*601*
Integration level	Connect+	Connect+	Connect+	Connect+
Supported firmware version	3.1.2.0392	3.1.2.0392	3.1.2.0392	3.1.2.0392
Automatic detection	√[4]	√[4]	√[4]	√[4]
Automatic provisioning	√	√	√	√
Automatic firmware upgrade	√	√	√	√
Max # of SIP registrations	2	2	3	6[1]
LCD screen size*	102x33	132x46	160x80	320x160
Available codecs**	1,2,3	1,2,3	1,2,3	1,2,3
Programmable hard keys				
Programmable soft keys	2	2	3	6
Offhook dialing	√	√	√	√
Offhook dialing & commit	√	√	√	√
One button message retrieval		√	√	√
Per-identity message retrieval	√	√	√	√
Per-identity MWI	√	√	√	√
Call lists – missed calls	√	√	√	√
Call lists – received calls	√	√	√	√
Call lists – placed calls	√	√	√	√
Handset operation	√	√	√	√
Headset operation	√	√	√	√
Speakerphone – full duplex	√	√	√	√
Call recording	√	√	√	√
Speed dial	√	√	√	√
PC Ethernet port	√[2]	√	√	√
Mute	√	√	√	√
Directory-enabled dialing	√	√	√	√
DND	√	√	√	√
PoE (Power over Ethernet)	√	√	√[1]	√
SRTP				
TLS	√[3]	√[3]	√[3]	√[3]
Backup proxy/registrar	√	√	√	√
Paging capable	√[5]	√[5]	√[5]	√[5]

* ^ (lines by characters); **x** (pixels)
** Codecs: 1 (G.711μ), 2 (G.711A), 3 (G.729A), 4 (G.722)
[1] With optional cable
[2] IP 330 – yes; IP 320 – single Ethernet port
[3] Not currently supported by NetVanta Unified Communications Server
[4] Phones in a different subnet than the UC server can also be automatically detected
[5] Supports unicast paging

Note: Polycom 301 no longer supported.

2.4 Grandstream

Capability	*BT101 /102*	*BT200 /201*	*GXP 280*	*GXP 1200*	*GXP 2000*	*GXP 2010*	*GXP 2020*
Integration level	Connect	Connect+	Connect+	Connect+	Connect+	Connect+	Connect+
Integration guide	**TN073**						
Supported firmware version	**1.1.0.3**	**1.1.6.16**	**1.1.6.27**	**1.1.6.16**	**1.1.6.16**	**1.1.6.16**	**1.1.6.16**
Automatic detection		√[3]	√[3]	√[3]	√[3]	√[3]	√[3]
Automatic provisioning		√	√	√	√	√	√
Automatic firmware upgrade		√	√	√	√	√	√
Max # of SIP registrations	1	1	1	4	4	4	6
LCD screen size*	12^	12^	128x32	130x64	130x64	130x64	320x160
Available codecs**	1,2,3,4,5,6,7	1,2,3,4,5,6,7	1,2,3,4,5,6,7	1,2,3,4,5,6,7	1,2,3,4,5,6,7	1,2,3,4,5,6,7	1,2,3,4,5,6,7
Programmable hard keys							
Programmable soft keys							
Off-hook dialing							
Off-hook dialing & commit							
One button message retrieval	√	√	√	√	√	√	√
Per-identity message retrieval							
Per-identity MWI							
Call lists – missed calls			√	√	√	√	√
Call lists – received calls	√	√	√	√	√	√	√
Call lists – placed calls	√	√		√	√	√	√
Handset operation	√	√	√	√	√	√	√
Headset operation	√	√	√	√	√	√	√
Speakerphone – full duplex	√	√	√	√	√	√	√
Call recording							
Speed dial				√	√	√	√
PC Ethernet port	√***	√	√	√	√	√	√
Mute	√	√	√	√	√	√	√
Directory-enabled dialing			√	√	√	√	√
DND			√	√	√	√	√
PoE (Power over Ethernet)				√	√	√	√
SRTP			√[1]	√[1]	√[1]	√[1]	√[1]
TLS							
Backup proxy/registrar			√[2]	√[2]	√[2]	√[2]	√[2]
Paging capable	√[4]	√[4]	√[4]	√[4]	√[4]	√[4]	√[4]

* ^ (digits); **x** (pixels); **"** (inches)
** Codecs: 1 (G.711μ/A), 2 (G.729A/B), 3 (G.723,1), 4 (G.726), 5 (G.722), 6 (GSM), 7 (iLBC)
*** BT101 does not have a PC Ethernet port
[1] Not currently supported by NetVanta Unified Communications Server
[2] DNS-SRV only
[3] Phones in a different subnet than the UC server can also be automatically detected
[4] Supports unicast paging

2.5 snom

Capability	300	320	360	370	820
Integration level	Connect+	Connect+	Connect+	Connect+	Connect+
Supported firmware version	7.3.14***	7.3.14***	7.3.14***	7.3.14**	8.1.3
Automatic detection	√[5]	√[5]	√[5]	√[5]	√
Automatic provisioning	√[1,2]	√[1,2]	√[1,2]	√[1,2]	√
Automatic firmware upgrade	√[4]	√[4]	√[4]	√[4]	√
Max # of SIP registrations	4	12	12	12	12
LCD screen size*	2^16	2^24	128x64	128x64	320x240
Available codecs**	1,2,3,4,5,6,7	1,2,3,4,5,6,7	1,2,3,4,5,6,7	1,2,3,4,5,6,7	1,2,3,4,5,6,7
Programmable hard keys		12	12	12	16
Programmable soft keys					4
Off-hook dialing	√	√	√	√	√
Off-hook dialing & commit	√	√	√	√	√
Single button message retrieval	√	√	√	√	√
Per-identity message retrieval		√	√	√	√
Per-identity MWI		√	√	√	√
Call lists – missed calls	√	√	√	√	√
Call lists – received calls	√	√	√	√	√
Call lists – placed calls	√	√	√	√	√
Handset operation	√	√	√	√	√
Headset operation	√	√	√	√	√
Speakerphone – full duplex	√	√	√	√	√
Call recording		√	√	√	√
Speed dial	√	√	√	√	√
PC Ethernet port	√	√	√	√	√
Mute	√	√	√	√	√
Directory-enabled dialing	√	√	√	√	√
DND	√	√	√	√	√
PoE (Power over Ethernet)	√	√	√	√	√
SRTP	√[3]	√[3]	√[3]	√[3]	√[3]
TLS	√[3]	√[3]	√[3]	√[3]	√[3]
Backup proxy/registrar					
Paging capable	√[6,7]	√[6,7]	√[6,7]	√[6,7]	√[6,7]

* ^ (lines by characters); **x** (pixels)
** Codecs: 1 (G.711µ), 2 (G.711A), 3 (G.729A), 4 (G.722), 5 (G.723.1), 6 (G.726), 7 (GSM 6.10)
****** Upgrades from v3, 4, 5, 6 (to v7) require a special upgrade path:***
http://wiki.snom.com/Firmware/V7/Update.
[1] snom phones with factory default settings require one manual reboot after the first time they are provisioned.
[2] Provisioning server polled for changes every 24 hrs relative to last reboot (phone won't reboot if changes are detected)
[3] Not currently supported by NetVanta Unified Communications Server
[4] Once firmware version 7.3.7 has been applied to the phone all future upgrades will be performed automatically
[5] Phones in a different subnet than the UC server can also be automatically detected
[6] Supports unicast paging
[7] Supports multicast paging

2.6 Aastra (67xxi series)

Capability	*6730i/6731i*	*6751i*	*6753i*	*6755i*	*6757i/ 6757iCT*
Integration level	Connect+	Connect+	Connect+	Connect+	Connect+
Supported firmware version	2.5.1	2.5.1	2.5.1	2.5.1	2.5.1
Automatic detection	√[1]	√[1]	√[1]	√[1]	√[1]
Automatic provisioning	√	√	√	√	√
Automatic Firmware Upgrade	√	√	√	√	√
Max # of SIP registrations	6	1	9	9	9
LCD screen size*	3-	3-	3-	144x75	144x128
Available codecs**	1,2,3,4	1,2,3	1,2,3	1,2,3	1,2,3
Programmable hard keys			4	6	
Programmable soft keys				6	12
Offhook dialing	√	√	√	√	√
Offhook dialing & commit	√	√	√	√	√
One button message retrieval		√	√	√	√
Per-identity message retrieval		√	√	√	√
Per-identity MWI					
Call lists – missed calls	√	√	√	√	√
Call lists – received calls	√	√	√	√	√
Call lists – placed calls	√	√	√	√	√
Handset operation	√	√	√	√	√
Headset operation		√	√	√	√
Speakerphone – full duplex	√	√	√	√	√
Call recording					
Speed dial	√	√	√	√	√
PC Ethernet port	√[4]	√	√	√	√
Mute	√	√	√	√	√
Directory-enabled dialing	√	√	√	√	√
DND	√	√	√	√	√
PoE (Power over Ethernet)	√	√	√	√	√
SRTP					
TLS					
Backup proxy/registrar	√	√	√	√	√
Paging capable	√[2,3]	√[2,3]	√[2,3]	√[2,3]	√[2,3]

* ^ (lines by characters); **x** (pixels); **-** (lines)
** Codecs: 1 (G.711µ), 2 (G.711A), 3 (G.729A), 4 (G.722 wideband)

[1] Phones in a different subnet than the UC server can also be automatically detected
[2] Supports unicast paging
[3] Supports multicast paging
[4] 6731i only; 6730i requires power adapter (included)

2.7 Aastra (9143i, 9480i, 9480iCT)

Capability	*9143i*	*9480i / 9480iCT*
Integration level	Connect+	Connect+
Supported firmware version	2.5.1	2.5.1
Automatic detection	√[1]	√[1]
Automatic provisioning	√	√
Automatic Firmware Upgrade	√	√
Max # of SIP registrations	3	4
LCD screen size*	3^16	8^20
Available codecs**	1,2,3	1,2,3
Programmable hard keys	7	
Programmable soft keys		6
Offhook dialing	√	√
Offhook dialing & commit	√	√
One button message retrieval	√	√
Per-identity message retrieval		√
Per-identity MWI		
Call lists – missed calls	√	√
Call lists – received calls	√	√
Call lists – placed calls	√	√
Handset operation	√	√
Headset operation	√	√
Speakerphone – full duplex	√	√
Call recording		
Speed dial	√	√
PC Ethernet port	√	√
Mute	√	√
Directory-enabled dialing	√	√
DND	√	√
PoE (Power over Ethernet)	√	√
SRTP		
TLS		
Backup proxy/registrar	√	√
Paging capable	√[2,3]	√[2,3]

* ^ (lines by characters); **x** (pixels); **-** (lines)
** Codecs: 1 (G.711µ), 2 (G.711A), 3 (G.729A)

[1] Phones in a different subnet than the UC server can also be automatically detected
[2] Supports unicast paging
[3] Supports multicast paging

2.8 Aastra (480i, 480iCT, 9112i, 9133i)

Capability	*9112i*	*9133i*	*480i / 480iCT*
Integration level	Connect+	Connect+	Connect+
Supported firmware version	**1.4.3**	**1.4.3**	**1.4.3**
Automatic detection	√[1]	√[1]	√[1]
Automatic provisioning	√	√	√
Automatic Firmware Upgrade	√	√	√
Max # of SIP registrations	1	3	4
LCD screen size*	3^16	3^16	8^20
Available codecs**	1,2,3	1,2,3	1,2,3
Programmable hard keys	2	7	
Programmable soft keys			18
Offhook dialing	√	√	√
Offhook dialing & commit	√	√	√
One button message retrieval	√	√	√
Per-identity message retrieval			√
Per-identity MWI			
Call lists – missed calls	√	√	√
Call lists – received calls	√	√	√
Call lists – placed calls	√	√	√
Handset operation	√	√	√
Headset operation	√	√	√
Speakerphone – full duplex	√	√	√
Call recording			
Speed dial	√	√	√
PC Ethernet port	√	√	√
Mute	√	√	√
Directory-enabled dialing	√	√	√
DND	√	√	√
PoE (Power over Ethernet)		√	√
SRTP			
TLS			
Backup proxy/registrar	√	√	√
Paging capable	√[2,3]	√[2,3]	√[2,3]

* ^ (lines by characters); **x** (pixels); **-** (lines)
** Codecs: 1 (G.711µ), 2 (G.711A), 3 (G.729A)

[1] Phones in a different subnet than the UC server can also be automatically detected
[2] Supports unicast paging
[3] Supports multicast paging

2.9 Cisco

Capability	*7912*	*7940*	*7960*
Integration level	Interoperable	Interoperable	Interoperable
Integration guide	TN078	TN077	TN077
Supported firmware version	8.0.1 (060412A)	POS3-08-9-00	POS3-08-9-00
Automatic detection		√[1]	√[1]
Automatic provisioning		√	√
Automatic firmware upgrade		√	√
Max # of SIP registrations	1	2	6
LCD screen size*	192x64	4.25^3	4.25^3
Available codecs**	1,2,3	1,2,3	1,2,3
Programmable hard keys			
Programmable soft keys			
Off-hook dialing			
Off-hook dialing & commit	√	√	√
Single button message retrieval	√	√	√
Per-identity message retrieval			
Per-identity MWI			
Call lists – missed calls	√	√	√
Call lists – received calls	√	√	√
Call lists – placed calls	√	√	√
Handset operation	√	√	√
Headset operation	√	√	√
Speakerphone – full duplex	√	√	√
Call recording			
Speed dial	√	√	√
PC Ethernet port	√	√	√
Mute	√	√	√
Directory-enabled dialing	√	√	√
DND	√	√	√
PoE (Power over Ethernet)			
SRTP			
TLS			
Backup proxy/registrar	√	√	√
Paging capable	√[2]	√[2]	√[2]

* ^ (inches); x (pixels)
** Codecs: 1 (G.711µ), 2 (G.711A), 3 (G.729A)

[1] Phones in a different subnet than the UC server can also be automatically detected
[2] Phone must be configured with an additional identity used solely for paging purposes

2.10 Mitel

Capability	*5220*	*5224*	*5235*	*5330*	*5340*
Integration level	**Interoperable**	**Interoperable**	**Interoperable**	**Interoperable**	**Interoperable**
Integration guide	**TN076**	**TN076**	**TN076**	**TN076**	**TN076**
Supported firmware version	**R7.1.07.01.00.09**	**R7.1.07.01.00.09**	**R7.1.07.01.00.09**	**R7.1.07.01.00.09**	**R7.1.07.01.00.09**
Automatic detection	√[3]	√[3]	√[3]	√[3]	√[3]
Automatic provisioning	√[1]	√[1]	√[1]	√[1]	√[1]
Automatic Firmware Upgrade	√	√	√	√	√
Max # of SIP registrations	12	12	12	12	24
LCD screen size*	2^20	2^20	340x240	160x320	160x320
Available codecs**	1,2,3	1,2,3	1,2,3	1,2,3,4	1,2,3,4
Programmable hard keys	24	24	24	24	48
Programmable soft keys					
Off-hook dialing	√	√	√	√	√
Off-hook dialing & commit					
One button message retrieval	√	√	√	√	√
Per-identity message retrieval					
Per-identity MWI					
Call lists – missed calls	√	√	√	√	√
Call lists – received calls	√	√	√	√	√
Call lists – placed calls	√	√	√	√	√
Handset operation	√	√	√	√	√
Headset operation	√	√	√	√	√
Speakerphone – full duplex	√	√	√	√	√
Call recording					
Speed dial	√	√	√	√	√
PC Ethernet port	√	√	√	√	√
Mute	√	√	√	√	√
Directory-enabled dialing			√	√	√
DND	√	√	√	√	√
PoE (Power over Ethernet)	√	√	√	√	√
SRTP	√[2]	√[2]	√[2]	√[2]	√[2]
TLS					
Backup proxy/registrar					
Paging capable	√[4]	√[4]	√[4]	√[4]	√[4]

* ^ (lines by characters); **x** (pixels)
** Codecs: 1 (G.711μ), 2 (G.711A), 3 (G.729A), 4 (G.722)

[1] Mitel phones must be switched manually to use SIP firmware
[2] Not currently supported by NetVanta Unified Communications Server
[3] Phones in a different subnet than the UC server can also be automatically detected
[4] Supports unicast paging

2.11 Linksys

Capability	*SPA941*
Integration level	Interoperable
Integration guide	TN079
Supported firmware version	5.1.8
Automatic detection	√[2]
Automatic provisioning	√
Automatic firmware upgrade	
Max # of SIP registrations	4
LCD screen size*	128x64
Available codecs**	1,2,3,4,5
Programmable hard keys	
Programmable soft keys	
Off-hook dialing	√
Off-hook dialing & commit	√
Single button message retrieval	√
Per-identity message retrieval	
Per-identity MWI	
Call lists – missed calls	√
Call lists – received calls	√
Call lists – placed calls	√
Handset operation	√
Headset operation	√
Speakerphone – full duplex	√
Call recording	
Speed dial	√
PC Ethernet port	√
Mute	√
Directory-enabled dialing	√
DND	√
PoE (Power over Ethernet)	√
SRTP	√[1]
TLS	
Backup proxy/registrar	√[1]
Paging capable	√[3]

* x (pixels)
** 1 (G.711 μ), 2 (G.711A), 3 (G.729A), 4 (G.723.1), 5 (G.726)

[1] Not currently supported by NetVanta Unified Communications Server
[2] Phones in a different subnet than the UC server can also be automatically detected
[3] Supports unicast paging

3 Software SIP Telephones

	CounterPath		SJ Labs
Capability	*X-Lite*	*eyeBeam*	*SJphone*
Integration level	Connect	Connect	Interoperable
Integration guide	TN080	TN030	TN082
Supported firmware version	3.0.41150	1.5.5	1.65.377a
Automatic detection			
Automatic provisioning			
Automatic firmware upgrade	√[1]	√[1]	
Max # of SIP registrations	1	10	<no limit>
Available codecs*	1,2,8,9,12,14, 15,16,17	1,2,3,7,8,9,10,11,12, 13,14,15,16,17	1,2,7,12
One button message retrieval	√[2]	√[2]	√
Per-identity message retrieval	√[2]	√[2]	√
Per-identity MWI	√[2]	√[2]	√
Call lists – missed calls	√	√	√
Call lists – received calls	√	√	√
Call lists – placed calls	√	√	√
Call lists – blocked calls	√	√	
Video support	√	√	
Call recording	√	√	
Speed dial			
Mute	√	√	√
Directory-enabled dialing	√	√	√
DND	√	√	√
SRTP	√[2]	√[2]	
TLS	√[2]	√[2]	
Backup proxy/registrar			
Paging capable	√[3]	√[3]	√[3]

* Codecs: 1 (G.711µ), 2 (G.711A), 3 (G.729A), 4 (G.722), 5 (G.723.1), 6 (G.726), 7 (GSM 6.10), 8 (Broadvoice-32), 9 (Broadvoice-32 FEC), 10 (DVI4), 11 (DVI4 Wideband), 12 (iLBC), 13 (L16 PCM Wideband), 14 (Speex), 15 (Speex FEC), 16 (Speex Wideband), 17 (Speex Wideband FEC)

[1] Optionally provided by CounterPath via their software upgrade system.
[2] Not currently supported by the UC server
[3] Phone must be in auto-answer mode

4 Conference SIP Telephones

4.1 Polycom SoundStation IP

Capability	4000	6000 (HD)	7000 (HD)
Integration level	Connect+	Connect+	Connect+
Supported firmware version	3.1.2.0392	3.1.2.0392	3.1.2.0392
Automatic detection	√[3]	√[3]	√[3]
Automatic provisioning	√	√	√
Automatic firmware upgrade	√	√	√
Max # of SIP registrations	1	1	1
LCD screen size*	248x68	248x68	255x128
Available codecs**	1,2,3	1,2,3,4	1,2,3,4
Programmable hard keys			
Programmable soft keys			
Offhook dialing	√	√	√
Offhook dialing & commit	√	√	√
One button message retrieval			
Per-identity message retrieval			
Per-identity MWI			
Call lists – missed calls	√	√	√
Call lists – received calls	√	√	√
Call lists – placed calls	√	√	√
Handset operation			
Headset operation			
Speakerphone – full duplex	√	√	√
Call recording			
Speed dial			
PC Ethernet port			
Mute	√	√	√
Directory-enabled dialing			
DND	√	√	√
PoE (Power over Ethernet)	√[1]	√	√
SRTP			
TLS	√[2]	√[2]	√[2]
Backup proxy/registrar	√	√	√
Paging capable	√[4]	√[4]	√[4]

* ^ (lines by characters); **x** (pixels)
** Codecs: 1 (G.711µ), 2 (G.711A), 3 (G.729AB), 4 (G.722, G.722.1, G.722.1C)

[1] With optional cable
[2] Not currently supported by the NetVanta Unified Communications Server
[3] Phones in a different subnet than the UC server can also be automatically detected
[4] Supports unicast paging

5 Video SIP Telephones

Capability	*Grandstream GXV3000*	*Polycom VVX 1500*
Integration level	**Connect+**	**Connect+**
Supported firmware version	**1.0.1.27**	**3.1.2.0750**
Automatic detection	√[1]	√[1]
Automatic provisioning	√	√
Automatic firmware upgrade	√	√
Max # of SIP registrations	3	6
LCD screen size*	5.6”	7” touch screen
Available audio codecs**	1,2,3,4,8,9,10	1,2,3,4,5,6
Available video codecs***	3	1,2,3
Programmable hard keys		
Programmable soft keys		6
Offhook dialing		√
Offhook dialing & commit		√
One button message retrieval	√	√
Per-identity message retrieval		√
Per-identity MWI		√
Call lists – missed calls	√	√
Call lists – received calls	√	√
Call lists – placed calls	√	√
Handset operation	√	√
Headset operation	√	√
Speakerphone – full duplex	√	√
Call recording		√
Speed dial		√
PC Ethernet port	√	√
Mute	√	√
Directory-enabled dialing		√
DND	√	√
PoE (Power over Ethernet)		√
SRTP	√[2]	
TLS		√[2]
Backup proxy/registrar	√[3]	√
Paging capable	√[4]	√[4]

* ^ (lines by characters); **x** (pixels)
** Audio codecs: 1 (G.711µ), 2 (G.711A), 3 (G.729AB), 4 (G.722), 5 (G.722.1), 6 (G.722.1C), 7 (G.723.1), 8 (G.726), 9 (GSM), 10 (iLBC)
*** Video codecs: 1 (H.263), 2 (H.263+ (1998), 3 (H.264)
[1] Phones in a different subnet than the UC server can also be automatically detected
[2] Not currently supported by NetVanta Unified Communications Server
[3] DNS-SRV only
[4] Supports unicast paging

6 SIP Wireless Servers

6.1 Polycom Kirk Wireless Server

Capability	*KWS300*	*KWS600v3*	*KWS6000*
Integration level	**Connect+**	**Connect**	**Connect+**
Integration guide	**TN106**	**TN112**	**TN113**
Supported firmware version	**PCS03A**	**PCS05Ac**	**PCS03A**
Automatic detection	√[7,8]		√[7,8]
Automatic provisioning	√		√
Automatic firmware upgrade	√		√
Radio Interface	DECT	DECT	DECT
Max # of Kirk Wireless Servers		1 – single-cell 256 – multi-cell	256[3]
Max # of registered handsets	12	35 – single-cell 1500 – multi-cell	4096[4]
Max # of repeaters	6	6 – single cell 1,2 or 3 – per 600v3 256[1] – whole system	6 per base station
Maximum # of simultaneous calls	4	12 – single-cell 11[2] – multi-cell	11 per base station 32 per Kirk Media Resource[5]
Available codecs*	1,2	1,3,4,5,6,7,8	1,2,8[6]
Supported handsets	3040, 4020, 4040, 4080, 5020, 5040	3040, 4020, 4040, 4080, 5020, 5040	3040, 4020, 4040, 4080, 5020, 5040
PoE (Power over Ethernet)	√	√	√
Paging capable			

* Codecs: 1 (G.711μ), 2 (G.711A), 3 (G.723), 4 (G.726), 5 (G.729), 6 (G729A), 7 (G729B), 8(G.729AB)
[1] Minus number of primary and secondary KIRK Wireless Servers
[2] No of KWS600v3 radios
[3] Including base stations and repeaters
[4] License stepwise scalable up to 4096
[5] Maximum number of Kirk Media Resources is 32 (using G.711)
[6] A codec module is required for G.729AB
[7] Kirk handsets require a manual step to register with the KWS server; see the corresponding Tech Note for details.
[8] The KWS300 and KWS6000 cannot both be deployed in the same network due to auto detection mechanism conflicts.

7 SIP Wireless Handsets

7.1 Polycom SpectraLink

Capability	*8002*
Integration level	Connect
Integration guide	**TN115**
Supported firmware version	**130.009**
Automatic detection	
Automatic provisioning	
Automatic firmware upgrade	
Max # of SIP registrations	5
Radio Interface	802.11b
Wireless Security	Standard WEP 40/128 bit WPA/WPA2 with Pre-Shared Key
Battery Life	Up to 3 hours talk time Up to 50 hours standby
Maximum # of simultaneous calls	1
Available codecs*	1,2
DND	√
Paging capable	

* Codecs: 1 (G.711μ), 2 (G.711A)

8 SIP Paging Speakers, Controllers, Amplifiers, and Intercoms

8.1 CyberData

Capability	*VoIP Speaker*	*VoIP Paging Server*	*VoIP Zone Controller*	*VoIP Paging Amplifier*	*VoIP Loudspeaker Amplifier*	*VoIP Intercom*
Integration level	**Connect**	**Connect**	**Connect**	**Connect**	**Connect**	**Connect**
Integration guide	TN084	TN084	TN085	TN086	TN086	TN103
Supported firmware version	4.0.5	1.22	1.07	3.05	3.05	2.04
Automatic detection						
Automatic provisioning						
Automatic firmware upgrade						
Provisioning protocols	TFTP	TFTP	TFTP	TFTP	TFTP	TFTP
Max # of SIP registrations	1	1	1	1	1	1
Available codecs*	1,2	1	1	1,2	1	1,2
Paging zones	10	100	4	10	10	
Paging zone groups			15			
PoE capable	√	√	√	√	√	√
Multicast capable	√	√		√	√	

* Codecs: 1 (G.711µ), 2 (G.711 a)

8.2 Atlas Sound

Capability	*VoIP Speaker I8S/I8SC/I8SCM/ IHVP/I128SYS*	*VoIP Zone Controller IPS-ZC1/IPS-ZC2*
Integration level	**Connect+**	**Connect+**
Supported firmware version	E286p54	E286p54
Automatic detection	√	√
Automatic provisioning	√	√
Automatic firmware upgrade	√	√
Provisioning protocols	TFTP	TFTP
Max # of SIP registrations	1	1
Available codecs*	1,2	1,2
Paging zones		1/2[1]
PoE capable	√	√
Multicast capable	√	√

* Codecs: 1 (G.711µ), 2 (G.711 a)

9 SIP/PSTN Gateways

9.1 Analog Gateways

Capability	Mediatrix *1204*	Quintum *Tenor AS & AF*	VegaStream *50 6x4*
Integration level	**Connect**	**Connect**	**Connect**
Integration guide	**TN024**	**TN037**	**TN026**
Environment			
Footprint (mm)	220 x 44 x 180	210 x 51 x 187	437 x 42 x 275
Power	110 VAC	110-240 VAC	100-240 VAC
LAN	10/100 BaseT	10/100 BaseT	10/100 BaseT
Rack mountable	√		√
Redundant power			
Redundant LAN			√
System			
Supported firmware version	5.0.15.92	P104-12-17	11.02.07.5R075S009
QoS	DiffServ, ToS, VLAN	DiffServ, ToS, VLAN	DiffServ, ToS, VLAN
Provisioning methods	TFTP, HTTP, Web	TFTP, HTTP, Web	TFTP, FTP, Web
Diagnostic	System logs	System logs	System logs, telnet
Number manipulation (In & Out)			
Number routing	√	√	√
SIP redundant			√
SIP interface			
Codecs*	1,2,4,5,6,7	1,2,4,6	1,2,4,6
Echo cancellation (msec)	32	Up to 128	0,8,16,32,64,128
T.38 fax	√	√	√
CNG Fax Tone detection		√	
SIP TCP	√	√	
Early media			
Call hold	√	√	√
Call transfer (blind)	√	√	√
Call transfer (supervised)	√	√	√
PSTN Interface			
Impedance	NA1, NA2	NA1, NA2	600, 900, CTR21
Caller ID	US, Canada	US, Canada	US
Line gain (dBm)	-30 ←→ 20	-30 ←→ 20	-14 ←→ 14
Answer/Disconnect supervision**			√
Number of trunks	4	2,4,6,8	4x6

9.2 Digital Gateways

Capability	AudioCodes *M1000*	Quintum *Tenor DX*	VegaStream *400*
Integration level	**Connect**	**Connect**	**Connect**
Integration guide	**TN027**	**TN069**	**TN028**
Environment			
Footprint	1U, 19" rack mount	1U, 19" rack mount	1U, 19" rack mount
Power	90-260 VAC	100-260 VAC	110-240 VAC
LAN	10/100 BaseT	10/100 BaseT	10/100 BaseT
Rack mountable	√	√	√
Redundant power	√		
Redundant LAN	√	√	√
System			
Supported firmware version	4.80A.033.004	P104-12-17	10.02.07.25R072S018
QoS	DiffServ, VLAN	DiffServ, ToS, VLAN	DiffServ, ToS, VLAN
Provisioning methods	BootP, TFTP, Web	BootP, TFTP, Web	TFTP, FTP, Web
Diagnostic	System logs	System logs	System logs, telnet
Number manipulation (In/Out)	√	√	√
Number routing	√	√	√
SIP redundant	√	√	√
SIP Interface			
Codecs*	1,2,4,5,6	1,2,4,5,6	1,2,4,6,7
Echo cancellation (msec)	32, 64	Up to 128	
T.38 fax	√	√	√
CNG Tone detection	√	√	
SIP TCP	√	√	√
Early media	√	√	
Call hold	√	√	√
Call transfer (blind)	√	√	√
Call transfer (supervised)	√	√	√
PSTN and Signaling			
Line coding	B8ZS, AMI, HDB3	B8ZS, AMI, HDB3	B8ZS, AMI, HDB3
Line framing	SF, ESF	SF, ESF	SF, ESF, CRC4, PCM30
Signaling	CAS: E&M, LS PRI: NI2, DMS100, 5ESS, QSIG	CAS: E&M, LS PRI: NI2, DMS100, 5ESS, QSIG	CAS: E&M Wink, Loopstart, Groundstart PRI: ATT, DMS, DMS M1, NI, QSIG
Trunk number (spans)	1,2,4	1,2,4,6,8	4
CallerID	√	√	√

10 Internet Telephony Service Providers (ITSPs)

Capability	BandTel	Bandwidth	Broadvox	AGN	babyTEL
Integration level	**Connect**	**Connect**	**Connect**	**Interoperable**	**Interoperable**
Integration guide	TN095	TN095	TN095	TN095	TN095
Ingate Startup Tool provisioning	√	√	√		
Ingate Settings					
Account Type provisioning	√		√		√
Dial Plan	√	√	√	√	√
Static Routing					
Network Routing					
Remote NAT Traversal					
DNS Name Resolution	√	√	√		√
ITSP SIP Redundant	√	√	√	√	
CPE SIP Redundant					
SIP Interface					
Registration	√			Optional	√
Digest Authentication (MD5)	√			√	√
Static Public IP Allocation		√	√	Optional	
Codecs	1,2,4,6	1,2,6	1,2,6	1,2,4,6	1,2,6
T.38 FAX	√”		√	√”	
DTMF (RFC 2833)	√	√	√	√	√
SIP over UDP	√	√	√	√	√
SIP over TCP		√			
Early Media	√			√	
Secure Media (SRTP)					
Transport Layer Security (TLS encryption)					
ITSP Services					
Inbound DID Calling	√	√	√	√	√
800 Inbound Calling	√	√	√	√	
911 and E911 Services	√	√		√	
ENUM (E.164) Number Delivery		√			
Incoming CallerID	√	√	√	√	√
Feature Interoperability					
Call Hold	√	√	√	√	√
Call Transfer	√	√	√	√	√
Conferencing					
Call Forwarding	√	√	√	√	√
Assisted Transfer	√	√	√	√	√
Music On Hold	√	√	√	√	√

* Codecs: 1 (G.711μ), 2 (G.711a), 3 (G.722), 4 (G.723.1), 5 (G.726), 6 (G.729a), 7 (G.729b), 8 (GSM 6.10)

Capability	VocalNet	Clarity Voice
Integration level	**Connect**	**Interoperable**
Integration guide	TN095	TN095
Ingate Startup Tool provisioning		
Ingate Settings		
Account Type provisioning	√	√
Dial Plan	√	√
Static Routing		
Network Routing		
Remote NAT Traversal		
DNS Name Resolution	√	√
ITSP SIP Redundant		
CPE SIP Redundant		
SIP Interface		
Registration		√
Digest Authentication (MD5)		√
Static Public IP Allocation		
Codecs	1,2,6	1,2,6
T.38 FAX		
DTMF (RFC 2833)	√	√
SIP over UDP	√	√
SIP over TCP		
Early Media		
Secure Media (SRTP)		
Transport Layer Security (TLS encryption)		
ITSP Services		
Inbound DID Calling	√	√
800 Inbound Calling	√	√
911 and E911 Services	√	√
ENUM (E.164) Number Delivery		
Incoming CallerID	√	√
Feature Interoperability		
Call Hold	√	√
Call Transfer Conferencing	√	
Call Forwarding	√	√
Assisted Transfer	√	
Music On Hold	√	√

* Codecs: 1 (G.711μ), 2 (G.711a), 3 (G.722), 4 (G.723.1), 5 (G.726), 6 (G.729a), 7 (G.729b), 8 (GSM 6.10)

11 SIP Firewalls / SIP ALG

Capability	Ingate	Intertex
Integration level	**Connect**	**Interoperable**
Integration guide	TN095	-
Models	**Firewalls:** 1190, 1500, 1550, 1590, 1900 **SIParators:** 19, 50, 59, 90	IX66, IX67, IX68
Supported Software Version	4.5.1	4.02
LAN/WAN Interface	10/100/1000 BaseT	10/100 BaseT
Rack Mountable	√	
Redundant LAN	√	
Firewall Functionality		
Stateful inspection	√	√
Packet filtering	√	√
DHCP Server & Proxy	√	√
Proxies for TCP, UDP and FTP	√	√
Flexible NAT and PAT footprint	√	√
SIP Functionality		
SIP Trunking with UC Server	√	
SIP Proxy	√	√
SIP Registrar	√	√
SIP B2BUA	√	√
SIP Traffic to Private IP addresses (NAT/PAT)	√	√
SIP over UDP	√	√
SIP over TCP	√	√
Transport Layer Security (TLS)	√	
Secure Media (SRTP descriptions) and Microsoft encrypted RTP	√	
Remote SIP Connectivity (Teleworker Support)	√	
Quality of Service (bandwidth limitation and prioritization)	√	
VPN Functionality		
X.509 certificates	√	√
PPTP server	√	√
IPsec (3DES, AES, NULL, MD5 and SHA1)	√	√