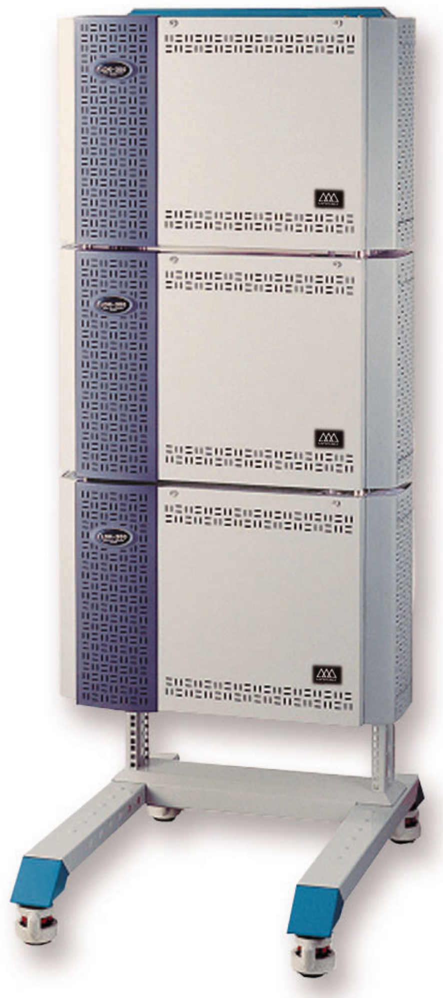


VODAVI XTS

IP Station

Product Sales Primer

Includes XTS*c*(Compact)

Information contained herein is subject to change.

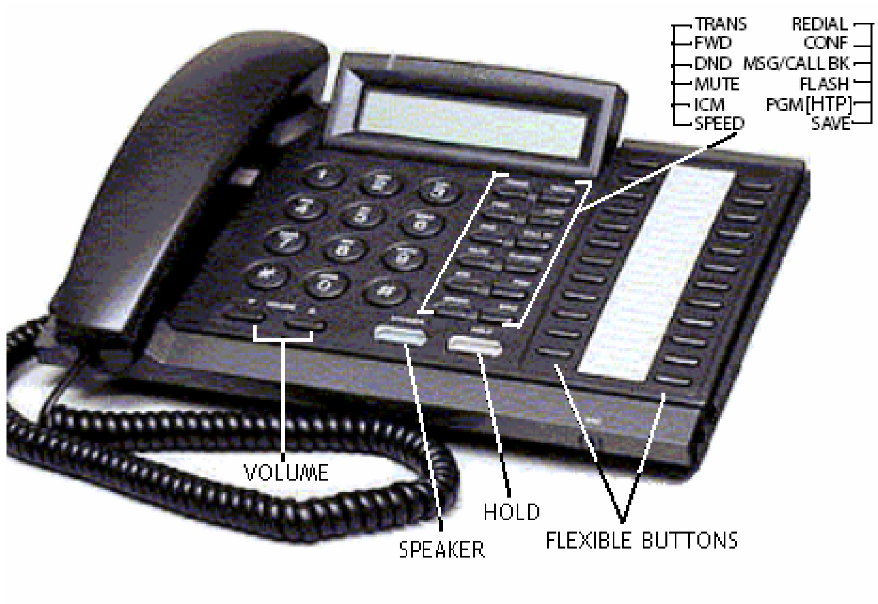
PRODUCT OVERVIEW

Vodavi's IP Solution for XTS and XTS*c* includes IP station support for users on wide-area networks, local networks, and broadband Internet services. By installing Voice over Internet Protocol (VoIP) cards and Vodavi IP telephones on the XTS system, the reach of the XTS office system can be extended to any place in the world where high-speed Internet or wide-area network service exists. Vodavi's implementation of Internet Protocol technology and engineering of the XTS system allows more than just telephone service over the network. An IP station becomes a full-featured XTS station, with almost every option and feature already available on standard digital stations.

FEATURES—IP-24DH Phone

Controls

The Vodavi IP phone is an IP digital handset with 24 flexible buttons and 16 fixed feature access keys (only 15 of the fixed keys are used with XTS).



Fixed Feature keys:

CONF – Conference	PGM – Tone Mode [H-T-P]
DND – Do Not Disturb	REDIAL
FLASH	SAVE
FWD – Call Forward	SPEAKER
HOLD	SPEED – Speed Dial functions
ICM – Not used	TRANS – Transfer
MSG/CALL BK – Message/Callback	VOLUME – up/down
MUTE	

All Flexible buttons and Fixed Feature buttons include a three-color LED to indicate the feature or circuit status. The default mapping for Flexible buttons matches the Vodavi digital station 24-button default map. The phone includes a 2-line, 48-character LCD, and a message wait indicator lamp. It features a full-duplex speakerphone and is headset compatible.

Calling Features

The full set of XTS station features is supported by the IP station, with the exception of these:

- Off-hook voice over (OHVO)
- Receiving pages (Can send pages)
- Background music (BGM)
- Selectable ring tones

Line and Power Interfaces

Voice packets are automatically prioritized to achieve maximum call quality. The IP-24DH has two 10/100 Base-T Ethernet ports, a "LAN" port and a "PC" port. An intelligent switch, which implements 802.1p protocol for voice packet priority, connects the two ports. This permits the network connection to be shared between the IP Phone and a desktop PC or other Fast Ethernet terminal. Both of the ports are auto-sensing to match the speed and duplex setting of the connected device. The phone can receive its power via an AC adaptor or from an Ethernet power source, such as a power-over-Ethernet switch.

Network Features

The IP phone can be configured as a DHCP client, so it can receive its local IP configuration from the network if the DHCP service is in use. It can also be configured manually, giving the installer full flexibility.

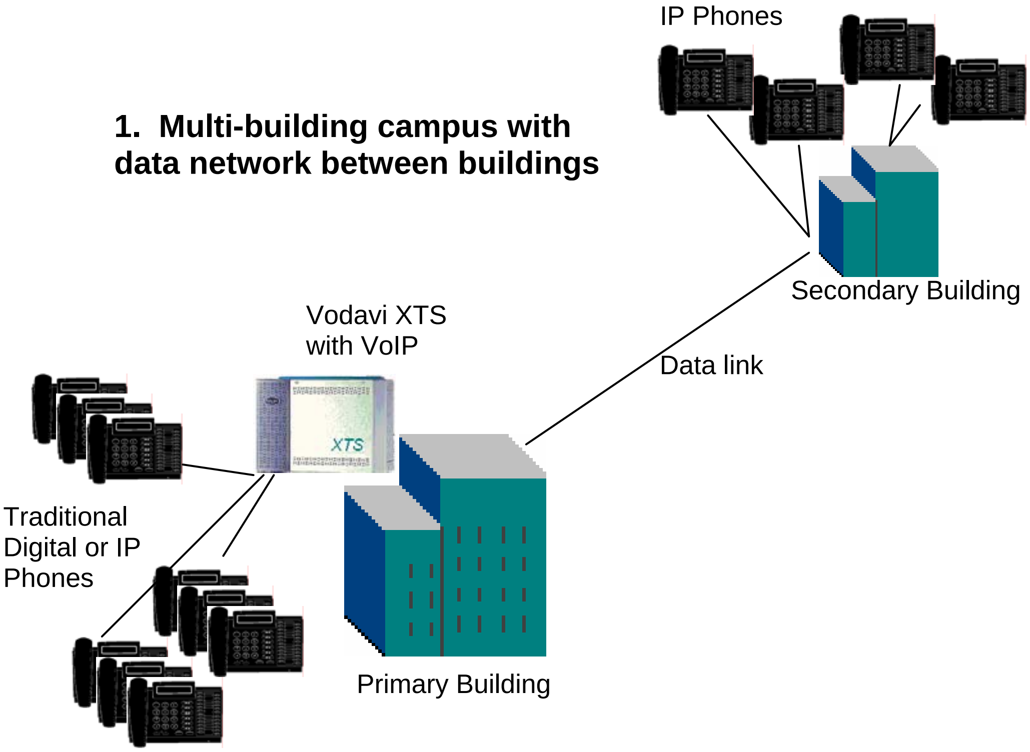
Device Options

The IP phone can be wall mounted using an optional wall mount bracket.

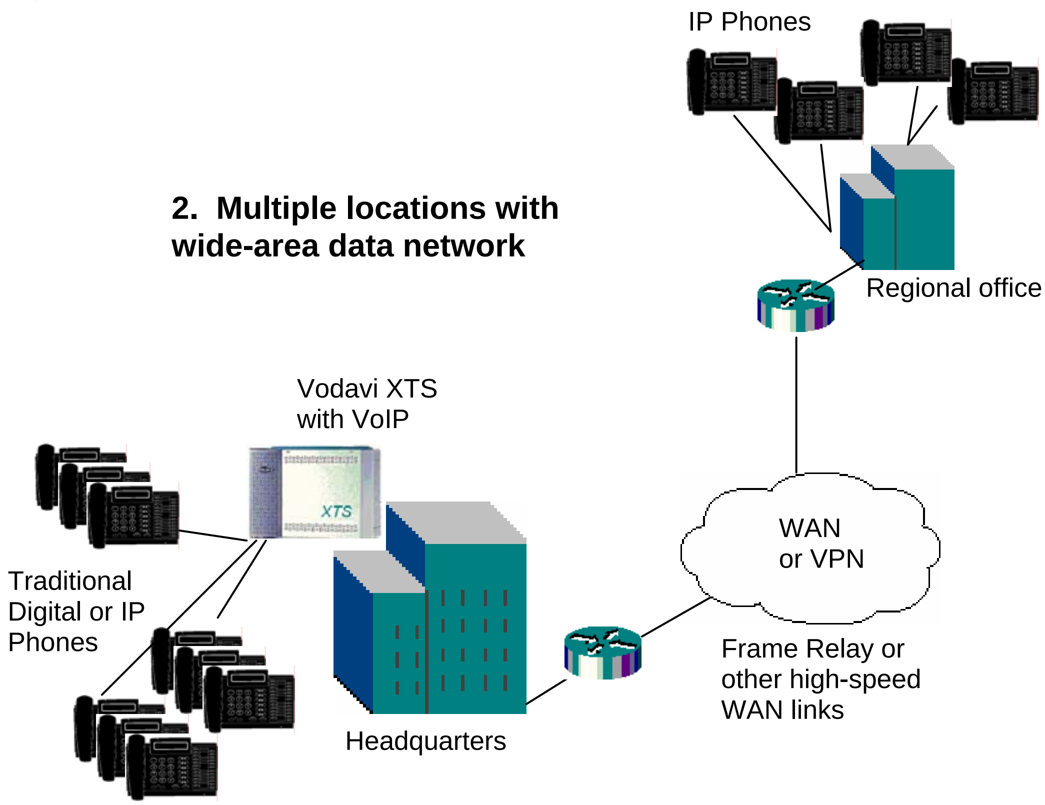
EXAMPLE APPLICATIONS

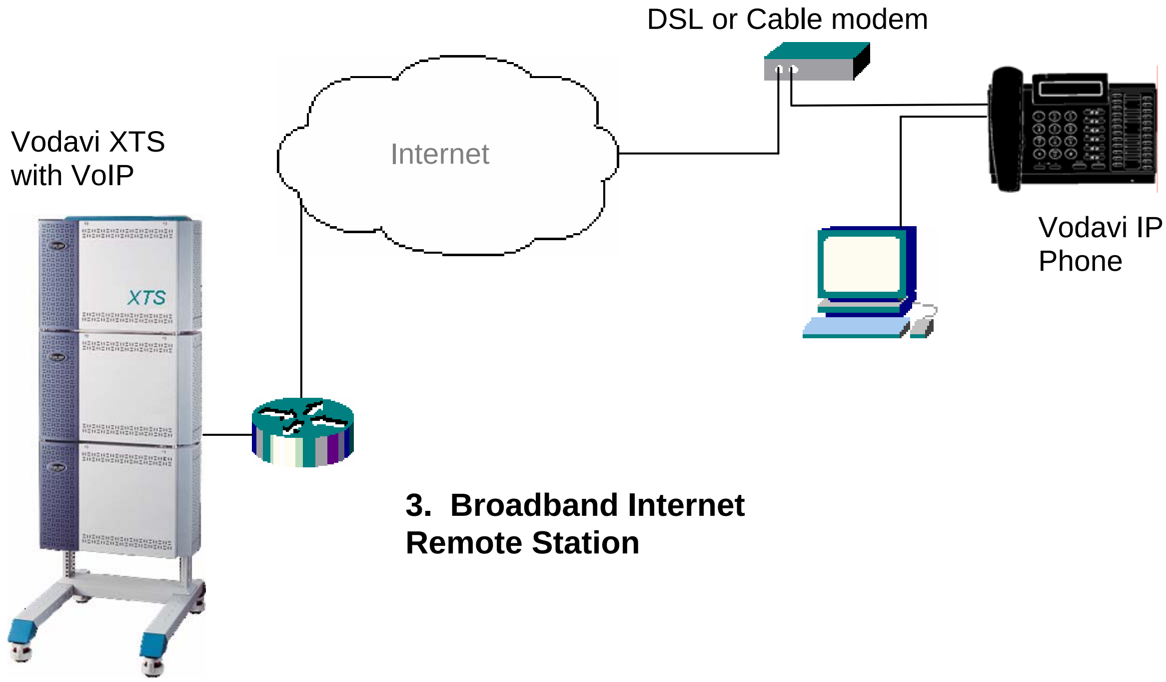
Suitable applications for IP phones include enterprises with multiple smaller locations, telecommuter offices, and distributed call center agents.

1. Multi-building campus with data network between buildings



2. Multiple locations with wide-area data network





3. Broadband Internet Remote Station

When using the Internet for access to the XTS system, quality of service cannot be guaranteed. The use of a hardware Virtual Private Network (VPN) may be necessary to ensure adequate voice quality and priority for voice traffic.

NOTE: When used as a remote station, E911 standards cannot be supported. The only outside lines available to the IP Phone are those on the XTS/c system. For emergency service, a separate analog telephone with a connection to the PSTN must be available.

The IP phone will go out of service if the connection to the Internet, WAN or LAN becomes unavailable.

INSTALLATION REQUIREMENTS

SYSTEM REQUIREMENTS

Software Levels

The IP phone requires an XTS/c system with at least one VoIP card. These are the minimum software levels for the system (Master Processor Board), the VoIP card, and the IP phone:

MPB1/MPB2	1.15p07
MPBE	1.02p07
XTSc	1.00
VoIP card	3.0Bb
IP Phone	1.2Ap

(current as of this writing—contact Vodavi Customer Service to verify whether newer versions are available.)

Port Usage

Provisioning

There must be at least one VoIP port available that can be allocated for the IP phone. To maintain system availability for the IP phone, it is best to maintain a one-to-one ratio between VoIP ports and IP phones.
There are two ways to allocate the resources on the VoIP card for IP Phones:

1. Reserved Resource Method
   In this method, a port is assigned to each remote phone in order to ensure that there will always be a connection available for that phone.

2. Pooled Resource Method
   This method allows multiple phones to share a pool of ports for remote access. This will work when there are multiple remote users but only a small percentage of them are actually in use at one time. This has the built-in possibility that a user may try to access dial tone and get no response from the system. If this occurs, add more VoIP ports to the system.

Reserved Access (non-blocking)

Number of VoIP Channels	Maximum Number of Phones Possible
2	2
4	4
6	6
8	8

Pooled Access (blocking)

Number of VoIP Channels	Maximum Number of Phones Possible
2	4
4	8
6	12
8	16

Using the pooled resource method, it is possible to over-allocate the IP phones, that is, to have an eight-port VoIP card with more than eight IP stations configured. For an IP phone to receive service when it goes off-hook, there must be an idle port on the VoIP card. Under-provisioning the system's VoIP ports can result in an IP station being denied a connection if all IP station ports are already in use.

System Usage

In the XTS VoIP programming, each VoIP port is counted against the total system CO port count. This applies even when a port is configured for use by an IP station.

NETWORK REQUIREMENTS

BANDWIDTH MINIMUMS-XTS SITE

Usage scenarios fall into one of two basic categories:

1. All phones and the XTS system are on the same network (Local, or LAN use)
2. The phones are not on the same network as the system (remote use)

In the first application, the phones and the system are in the same LAN. A typical 100-Mbps Fast Ethernet LAN will be provide sufficient bandwidth, and should be considered the minimum LAN speed to support IP stations.
In the second application, the IP phones are either on the Internet, or at some other location with a WAN connection to the XTS system. Connections to remote offices, the Internet, or WAN locations can vary widely. These are the minimums which should be used for the XTS's WAN/Internet connection:

Minimum and recommended bandwidths at G.711 vocoder setting:

	2 phones	4 phones	6 phones	8 phones
Recommended	512K	640K	T1	T1
Minimum	256K	256K	768K	1024K

The above table is for provisioning only the VoIP traffic through the connection. Keep in mind that if a site will have both voice and data using the same connection, much more bandwidth will be required. If a remote location will support more than one IP station, the above chart can also be used to determine the amount of network bandwidth required based on the number of IP stations.

The selection of the G.723.1 high-compression vocoder will reduce the amount of bandwidth required. The following table shows the difference in usage per call between using the G.711 and the G.723.1 vocoder.

NOTE: These figures only represent the amount of bandwidth taken up by VoIP calls—they are not recommended bandwidth minimums:

Vocoder	Bandwidth per call
G.711	110 Kbps
G.723.1	22 Kbps

Number of calls →→	2	4	6	8
Total B/W used at G.711*:	**220**	**440**	**660**	**880**
Total B/W used at G.723.1*:	**44**	**88**	**132**	**176**

* in Kbps

BANDWIDTH MINIMUMS-REMOTE STATION SITE

Vodavi recommends a broadband connection of at least 256Kbps at the remote site, if one IP phone is used. This minimum speed applies to both the upload and download rates. Although the phone uses up only 22 Kbps or 110 Kbps by itself, some bandwidth must remain available for data. More bandwidth generally gives a higher-quality connection.

Important: The bandwidth recommendations here are dependent upon the other network factors described on the next page being kept within acceptable limits.

OTHER NETWORK FACTORS

Latency, jitter, and packet loss

Latency (packet delay), jitter, and packet loss are measurable factors which must be kept within certain limits. Excessive amounts of any of these will have a detrimental effect on call quality. These factors are not constant on most networks, including the Internet. They can be controlled on a private, managed network.

Parameter	Acceptable value	Description
Latency (also called delay)	Less than 120 milliseconds	The duration it takes for a packet to move from one endpoint to another on the network. In this case, the endpoints are the IP phone and the IP-enabled XTS system.
Jitter	Less than 20 milliseconds	The variation in latency. When packets arrive at varying speeds, it can be disruptive to getting a clear audio stream.
Packet loss	Less than 2%	When some packets do not arrive, there are a gaps in the audio stream, leading to poor audio. Because voice is a real-time application, the missed packets will not be retransmitted.

You can employ some basic tests to measure the above parameters on the network. If the values are too high to support voice over IP, a network administrator or engineer may be able to improve them.

Network Address Translation

Network Address Translation, or NAT, involves using one IP address for internal communication, and another for Internet or WAN communication. Usually, the internal addresses used are private-class addresses. A router which performs NAT is able to re-label all of the packets so that the inside address is not the one seen by the outside host. This is done for two primary reasons:

1. The scarcity and expense of Internet-routable addresses; and
2. The relative safety of obscuring the true internal addressing scheme for a network

While this approach is great for providing Internet access to PCs with private-class addresses, it can cause problems for Voice Over IP. There are ways to overcome the issues that NAT causes for H.323-based VoIP systems. Certain router products are H.323-compliant, which means that with the correct programming they can support VoIP.

Vodavi's IP phones are capable of operating from behind a NAT device, such as a broadband home-office router. The XTS VoIP card may **not** be located behind a NAT, but must instead have a public, routable IP address.

Firewalls

A firewall is a network device or a software program on a router or server. Its purpose is to filter access into and out of the local network. It does so based on a rule set, allowing only certain network protocols and certain hosts to communicate through it. Many firewalls also perform Network Address Translation as well.

Vodavi recommends avoiding firewalls whenever possible. If a firewall must be involved, a firewall engineer or administrator should be consulted prior to installation so that problems can be minimized. The 'Protocols' section of this guide contains IP port information needed by those who intend to engineer a solution involving a firewall.

PART NUMBERS

VoIP Card	3037-0x where X represents IP Port count
Software – VoIP card	3037-01
IP Phone Kit	3814-02 includes keyset and AC power supply
IP Phone wall mount	3813-00 optional
Software—XTS system	3020-01
Documentation	Included with VoIP card for XTS

TECHNICAL DATA

STANDARDS

The Vodavi IP phone is capable of using an IP network such as a Local Area Network (LAN), the Internet, or a Wide Area Network (WAN) via its Fast Ethernet interface. The phone employs H.323 Revision 2 protocols and ensures Quality of Service (QoS) with the 802.1p/Q protocol standard. This protocol helps reduce echo, jitter, and latency. "Toll Quality" can be achieved under ideal network conditions.

The vocoders employed to digitize and packetize voice are G.711U-law, G.711A-law, and G.723.1. The vocoder is selectable by the installer configuring the VoIP card to achieve the goals of bandwidth conservation and best voice quality.

There are two options for powering the IP phone—via an AC power adaptor, or via an Ethernet device supporting the Power-over-Ethernet standard.

SPECIFICATIONS

IP Telephone (IP-24DH)

Dimensions and Weight

Height: 86.4mm/3.4in
Width: 269mm/10.6in
Depth: 196mm/7.7in
Weight: 0.92kg/2.0lb

AC Wall Adaptor Power

AC Voltage Input: 100-240 VAC, +/- 10% @50/60Hz
AC Power: 0.2 amps
DC Output Voltage: 48 VDC @ 0.1 amps

Ethernet Inline Power

802.3af Ethernet Power (Does not include IP-24D phone)
48 VDC over RJ-45 pins 4 & 5 (+), 7 & 8 (-) (includes IP-24D)

IP Phone Network Cable

A Category 5 or higher network cable is required for connecting the IP phone to a Fast Ethernet switch or router.

PROTOCOLS

IP Telephony Standard Protocol

H.323 revision 2 for call control

Vodavi IP Phone Protocol

For Vodavi advanced keyset functions

Vocoders

G.711u-law

G.711a-law

G.723.1

Network-Fast Ethernet w/Auto Negotiation

IEEE 802.3u

Voice Packet Prioritization

IEEE 802.1p/Q

IP Port numbers

H.323 PROTOCOLS	PORT	TRANSPORT	DESCRIPTION
H.323/H.225	1720 / 8057	Static TCP	H.323 Call Setup
H.323	1731	Static TCP	Audio Call Control
H.323/H.245	1024 – 65535	Dynamic TCP	Call Parameters
RTP	1024 – 65535	Dynamic UDP	Audio Stream Data
RTCP	1024 – 65535	Dynamic UDP	Control Information

IP PHONE PROTOCOL	PORT	TRANSPORT	DESCRIPTION
IPKTS Unicast	5588	Static UDP	System commands, polling
IPKTS Multicast	6254	Static UDP	System commands, polling

CAPACITIES

The number of IP phones which can be supported is limited by the number of VoIP cards installed in the XTS, and the number of available CO and station timeslots on the system. The number of VoIP cards accommodated by XTS varies with the type of main processor board (MPB1, MPB2, MPBE, XTSc) installed, and the number of cabinets.

Up to sixteen IP stations can be configured per eight-port VoIP card. This is underprovisioning using the pooled resource method, as described above.

Each VoIP card counts as eight CO lines, regardless of the number of daughter boards or VOIP ports in use. In addition, each IP phone configured on the system counts as a station.

To calculate how this will affect the system port count, multiply the number of VoIP cards by eight, then add the number of IP stations configured.

Examples:

VoIP cards: 1	CO ports used=	8
IP phones: 6	STA ports used=	6
	Total system ports used:	**14**

VoIP cards: 2	CO ports used=	16
IP phones: 16	STA ports used=	16
	Total system ports used:	**32**

The total system port limit still applies.

