UTS

UNIFIED TELEPHONE SYSTEM

Powered by Weltel Telecoms

Weltel Telecoms

UT-hosted

Cloud Business Voice | Internet & Networks | Print Solutions | Security Solutions | Tech Support

Unified Telephone Systems is based on one of the worlds most powerfull telephony platforms

FLEXIBLE/ROBUST/CUSTOMER FOCUSED

With UTS our core focus is development that is customer focused, our aim is to drastically reduce the cost of ownership and delivering a noticeable return on investment that no proprietary solution can match.

Unified Telephone Systems works for SMBs, Corporate Enterprises, Government and Contact Centers, offering robustness, advanced features and absolute flexibility and scalability.

Unified Telephone Systems

UT- *hosted*

Unified Telephone Systems

UTS is the product of ten years of work by a dedicated team of engineers. UTS is a proprietary and semi open source PBX that provides the same functionality as high end business telephone systems. It is the most flexible and scalable telephone system on the market, providing a broad array of features that are not yet available in even the most advanced competitive PBX systems.

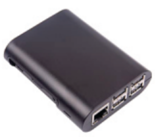
The software runs on Linux operating system, which also makes it one of the most secure telephone system on the market today!

Weltel Telecoms provides solutions to improve organisational efficiency by implementing information and communication systems.

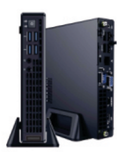
We offer expert support and advice, throughout South Africa. The UTS Telephone Systems has been fully implemented by Weltel Telecoms in a number of state and private organisations,offering superb functionalities, with no licensing costs.

We have a full team of in-house accredited Master engineers!

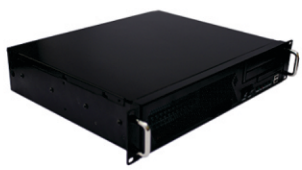
UT- *10*



UT- *50/100*



UT- *Ent*



UTS

UNIFIED TELEPHONE SYSTEM

Powered by Weltel Telecoms

Weltel Telecoms

FEATURES

- Voicemail to email
- MS Outlook integration
- Auto Attendant
- Call recording
- Name dialing
- Call reporting
- Music/Message on hold
- Automatic Route Selection
- Remote Extensions
- Call Distribution
- Flexibility and scalability

BENEFITS

- No Licensing costs
- Significant call savings
- Improved SLA levels
- Enhanced productivity
- Supports the use of existing infrastrucure in order to maximise cost saving

UT- hosted

Unified Communications

UTS offers an cross-platform Instant Messaging client optimized for businesses and organizations. It features built in support for group chat, telephony integration, and strong security. It also offers a great end-user experience with features like presence, group chat and tabbed conversations. It is written in Java and can be used standalone or as an add-on or plugin to certain Web browsers. Combined with the Openfire server, Spark IM is the easiest and best alternative to using unsecure public Instant Messaging networks.

Openfire supports the following features:

- Web-based administration panel
- Plugin interface
- Customizable
- LDAP connectivity
- User-friendly web interface and guided installation
- Database connectivity to store messages and contacts

UTS or proprietary?

The main scope of any organisation choosing to migrate from their existing system to a full UTS solution rests within the comfort and reassurance of obtaining full reliable coverage throughout its site. This would ensure an excellent Return on Investment when considering the following short and long term benefits:

- Excellent system integrity
- System architecture scalability
- Extensive features and call-centre functionalities
- Future-proof technology with reduced upgrades and no licensing costs.

Further scope can be found within the advantages of streamlining an existing phone system into a single IP-based reliable and cost-effective telephony solution, which would offer extensive scalability and could be further expanded to unlimited coverage.

OTHER SERVICES

Consultancy

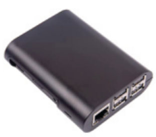
System Design

Project Management

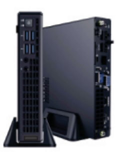
Procurement and deployment

Expert Advice

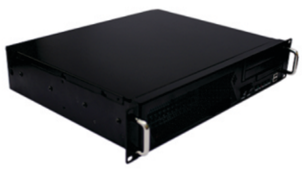
UT- 10



UT- 50/100



UT- Ent



UTS

UNIFIED TELEPHONE SYSTEM

Powered by Weltel Telecoms

Weltel Telecoms

UT-hosted

Cloud Business Voice | Internet & Networks | Print Solutions | Security Solutions | Tech Support

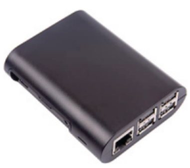
UTS™ Features

UTS telephony solutions offer a rich and flexible feature set. UTS offers both classical PBX functionality and advanced features, and interoperates with traditional standards-based telephony systems and Voice over IP systems.

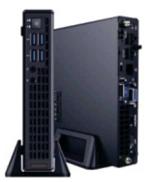
Call features

- ADSI On-Screen Menu System
- Alarm Receiver
- Append Message
- Authentication
- Automated Attendant
- Blacklists
- Blind Transfer
- Call Detail Records
- Call Forward on Busy
- Call Forward on No Answer
- Call Forward Variable
- Call Monitoring
- Call Parking
- Call Queuing
- Call Recording
- Call Retrieval
- Call Routing (DID & ANI)
- Call Snooping
- Call Transfer
- Call Waiting
- Caller ID
- Caller ID Blocking
- Caller ID on Call Waiting
- Calling Cards
- Conference Bridging
- Database Store / Retrieve
- Database Integration
- Dial by Name
- Direct Inward System Access
- Distinctive Ring
- Distributed Universal Number Discovery (DUNDi™)
- Do Not Disturb
- E911
- ENUM
- Fax Transmit and Receive (3rd Party OSS Package)
- Flexible Extension Logic
- Interactive Directory Listing
- Interactive Voice Response (IVR)
- Local and Remote Call Agents
- Macros
- Music On Hold
- Music On Transfer
  - Flexible Mp3-based System
  - Random or Linear Play
  - Volume Control
- Predictive Dialler
- Privacy
- Open Settlement Protocol (OSP)
- Overhead Paging
- Protocol Conversion
- Remote Call Pickup
- Remote Office Support
- Roaming Extensions
- Route by Caller ID
- SMS Messaging
- Spell / Say
- Streaming Media Access
- Supervised Transfer
- Talk Detection
- Text-to-Speech (via Festival)
- Three-way Calling
- Time and Date
- Transcoding
- Trunking
- VoIP Gateways
- Voicemail
  - Visual Indicator for Message Waiting
  - Stutter Dial tone for Message Waiting
  - Voicemail to email
  - Voicemail Groups
  - Web Voicemail Interface
- Zapateller

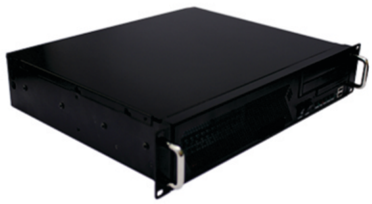
UT- 10



UT- 50/100



UT- Ent



UT- hosted

UTS

UNIFIED TELEPHONE SYSTEM

Powered by Weltel Telecoms

Weltel Telecoms

UT-hosted

Cloud Business Voice | Internet & Networks | Print Solutions | Security Solutions | Tech Support

System Capacity

UT-10: Up to 20 users
UT-50: Up to 50 users
UT-100: Up to 300 users
UT- Ent: Up to 1000 users

UT- hosted: Depending on connectivity max 30 users

Computer-Telephony Integration

- AGI
- Graphical Call Manager
- Outbound Call Spooling
- Predictive Dialler
- TCP/IP Management Interface

Scalability

- TDMoE (Time Division Multiplex over Ethernet)
  - Allows direct connection of legacy Hybrid PBX
  - Zero latency
  - Uses commodity Ethernet hardware
- Voice-over IP
  - Allows for integration of physically separate installations
  - Uses commonly deployed data connections
  - Allows a unified dialplan across multiple offices

Codecs

- ADPCM
- G.711 (A-Law & ! -Law)
- G.723.1 (pass through)
- G.726
- G.729 (through purchase of commercial license through Digium)
- GSM
- iLBC
- Linear
- LPC-10

Protocols

- IAX™ (Inter-Asterisk Exchange)
- H.323
- SIP (Session Initiation Protocol)
- MGCP (Media Gateway Control Protocol)
- SCCP (Cisco® Skinny®)

Traditional Telephony Interoperability

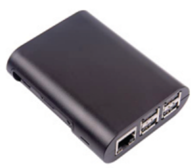
- E&M
- E&M Wink
- Feature Group D
- FXS
- FXO
- GR-303
- Loopstart
- Groundstart
- Kewlstart
- MF and DTMF support
- Robbed-bit Signaling (RBS) Types

PRI Protocols

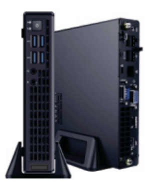
- 4ESS
- BRI (ISDN4Linux)
- DMS100
- EuroISDN
- Lucent 5E
- National ISDN2

UT- hosted

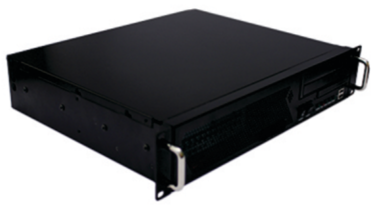
UT- 10



UT- 50/100



UT- Ent



UTS

UNIFIED TELEPHONE SYSTEM

Powered by Weltel Telecoms

Weltel Telecoms

Features

Playback and recording of audio in multiple formats, including HD formats.

DTMF detection and collection.

ODBC database access.

Web service access.

LDAP data access.

Calendaring data access.

Speech synthesis and recognition (requires 3rd party add-on components)

Branching and logic operations (if/then, while, for/next, etc.).

Benefits

Improve customer interactions by providing 24 hour access to basic automated services.

Reduce payroll costs or Repurpose employees to handle more valuable tasks.

UT- *hosted*

UTS As IVR Server

Description

IVR or Interactive Voice Response is the art of automating routine and repetitive communication tasks that would otherwise be serviced by operators, agents or other employees. The most frequently cited example of IVR is the “bank-by-phone” application offered by many banks. Other examples include automated night confrmations, service activations, credit card payments, and even call routing (often referred to as “automated attendant”).

IVR saves businesses money by handling tasks that would otherwise take the time and attention of a human. IVR applications receive input from the caller as digits (using the telephone keypad) or using speech recognition. Most IVR systems connect with a remote data source like a relational database, corporate directory or web service.

Traditional IVR systems are built on top of expensive proprietary voice engines which in turn are built on expensive proprietary telephony hardware. Asterisk simplifes the process of building an IVR and reduces the costs signifcantly. UTS’s dialplan scripting language includes commands to play recorded prompts, to collect digits or spoken responses and to reply with synthesized or recorded responses. Just as importantly, the dial plan language incorporates commands for reading from and writing to a number of data sources including databases, web services, LDAP directories and calendaring data stores.

The UTS voice engine is open source and is available free of charge to all.

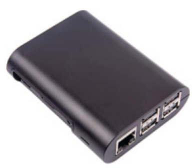
UTS-based IVR systems that connect over VoIP require no special hardware or software licenses.

Supported Scenarios

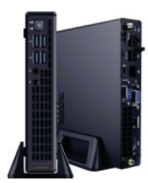
Network IVR connected directly to PSTN or VoIP trunks

Behind The Switch, connected to a PBX via Volp using legacy technologies

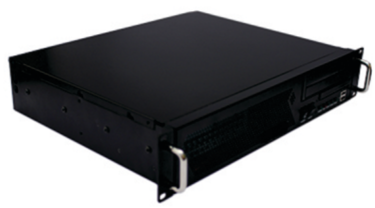
UT- *10*



UT- *50/100*



UT- *Ent*



UTS

UNIFIED TELEPHONE SYSTEM

Powered by Weltel Telecoms

Weltel Telecoms

- **Features**
  Support for the most common protocols including: SIP, IAX2, H.323
- **Dynamic call routing for** cost savings (“least cost routing”) or for redundancy/fail-over.
- Call data forwarding and manipulation across various technologies.
- Inexpensive solution based on commodity computer hardware.
- Based on stable and widely tested Linux operating system.
- **Benefits**
  Extend the life of legacy investments by adding IP telephony capabilities and services.
- Save money on long distance and international charges by using low cost VoIP services.
- Save money on toll charges by implementing private connections over the corporate WAN.
- Rapid return on investment (ROI) – generally less than 12 months.

UT- hosted

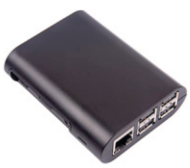
UTS As VoIP gateway

UTS can be used to build a gateway using a standard computer and one or more telephony interface cards. Alternately, UTS offers a line of turnkey gateways built using UTS. In either case, the end product is significantly more flexible and significantly less expensive than legacy gateway products. UTS's modular, multi-protocol architecture is particularly well suited to building gateways.

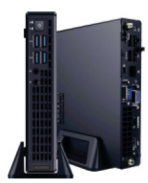
What Is A VoIP Gateway?

A VoIP gateway is used to build a bridge between the worlds of legacy telephony and the VoIP. Gateways are typically used to connect legacy phone sysetms (PBXs or ACDs) with VoIP resources, or to connect modern VoIP phone systems with legacy phone lines. Adding VoIP to a legacy PBX system is a great way to add features and reduce costs. The gateway connects to the legacy system through either analog or digital trunk ports. The PBX sees the gateway as either the phone company or as another networked PBX. Calls from the PBX to the outside world are converted into VoIP calls and sent over the Internet to a VoIP service provider or other VoIP peer. Calls coming from VoIP sources are converted into the appropriate legacy protocol and delivered to the PBX.
Using a gateway to connect a VoIP phone system to traditional phone lines makes sense in situations where SIP trunks are not available or where your application requires the reliability of the PSTN. It also makes it easy to build redundant systems: the gateway normally communicates with a primary IP PBX. In the event of a failure on the primary the gateway can communicate with a backup system. Other uses for VoIP gateways include staged migrations, where the gateway acts as a bridge between the PSTN, a legacy PBX and and a new IP PBX. In this case, the PSTN trunks are connected to one interface on the gateway. Another interface connects to the trunk port on the legacy PBX. The new IP PBX is integrated over a VoIP protocol (generally SIP). The gateway directs some incoming calls to the legacy PBX and others to the IP PBX. It also passes calls between the two PBXs.
This allows some department or other subdivisions of the company to remain on the legacy system while others move to the IP system.

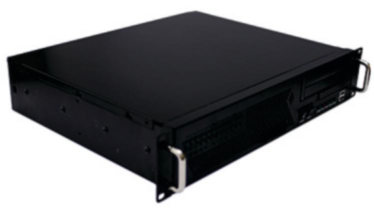
UT- 10



UT- 50/100



UT- Ent



UTS
UNIFIED TELEPHONE SYSTEM

Powered by Weltel Telecoms

Weltel Telecoms

Cloud Business Voice | Internet & Networks | Print Solutions | Security Solutions | Tech Support

Conferencing allows companies to save significantly on travel costs. In-person meetings are costly and time consuming. A conference bridge system can pay for itself in just one avoided "on-site" meeting.

Conferencing is the core of collaboration and enables distributed or virtual teams. Combined with VoIP connectivity for remote workers, conferencing makes it simple and affordable for a team to function across a diverse geography.

Weltel Group.

Tel: 0860 994 075
Mail: info@weltel.co.za

UT-hosted

UTS As A Conference Bridge

UTS includes a standard application called Conf Bridge. Conf Bridge is a high definition-capable conference bridge component that makes it easy to build stand-alone conferencing services or to integrate conferencing into other solutions, including IP PBX systems.

Creating a conference room is trivial, requiring only a few lines of Dial plan script. Conf Bridge includes a wealth of administrative features (mute participants, add / remove callers, etc.) and a rich event structure that allows developers to build fully integrated user interfaces. Conf Bridge also supports basic video conferencing, though this feature is currently considered experimental.

What Is A Conference Bridge?

A conference bridge allows a group of people to participate in phone call. The most common form of bridge allows participants dial into a virtual meeting room from their own phone. Meeting rooms can hold dozens or even hundreds of participants. This is in contrast to three-way calling, a standard feature of most phone systems which only allows a total of three participants. For most phone systems, conference bridging is an add-on feature that costs thousands of dollars.

Key Features

Conferencing systems typically support multiple conference rooms, each of which can contain multiple participants. The total number of rooms and participants varies depending on the model, hardware capabilities and licensing terms.

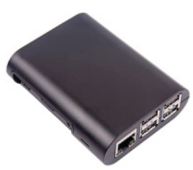
Most conference bridge systems allow the administrator to assign DID numbers to conference rooms. In some cases as single DID number connects callers with an IVR application that prompts for a room number.

Conference rooms can optionally be secured by a PIN number. Some systems use a common PIN for all participants while others use custom PINs for each.

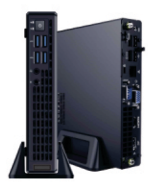
Advanced conferencing systems include a graphical user interface that allows all participants to see who is currently speaking and optionally who has joined the conference. Admnistrators and moderators generally have a more comprehensive view that includes advanced controls.

Some conference bridge systems include dynamic meeting rooms: rooms that are created on a scheduled basis. This is particularly common for larger systems where capacity planning is an issue.

UT- 10



UT- 50/100



UT- Ent