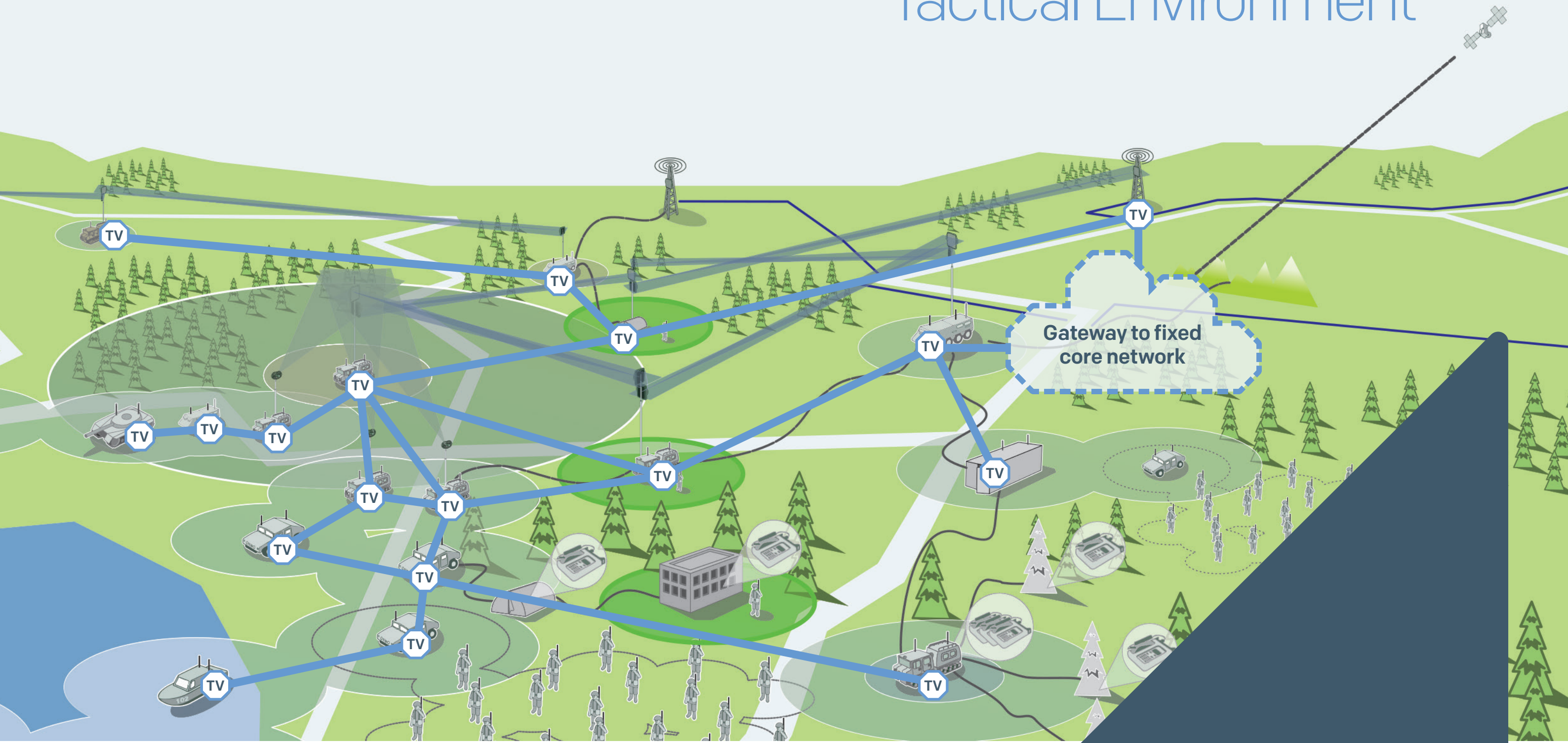
Bittium Tough VoIP Service™

Distributed VoIP Service Network for Tactical Environment



TV - Bittium Tough VoIP Service node

Bittium Tough VoIP Service™ network creates a highly resilient, distributed digital voice service in a tactical network. Tough VoIP Service can rapidly adapt to changes in the network. Network can be split or merged while maintaining the service available for all clients that are accessible. When network islands merge with the network, the clients will have core network connection automatically.

Tough VoIP Service is targeted for tactical level users and the call functions have been optimized for tactical networks. Call functions include for example distributed group calls, PTT calls, call prioritization (MLPP) and broadcast calls. Provided affiliation and registration service allows user mobility and affiliation of any phone number with any client without network administration.

Benefits

- Native support for MANET
- Zero configuration
- Seamless connectivity across tactical and fixed networks
- Easy integration to other SIP based VoIP services and phones

Use case examples

- Tactical VoIP network in the battlefield
- Bridges different network technologies and vendor networks

FOR MORE INFORMATION, PLEASE CONTACT:
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Bittium Tough VoIP Service™
Technical Features

Features

SUPPORTED VOIP CALLS

- Calls within multiple Tough VoIP Service nodes
- Calls between two different Tough VoIP Service networks
- Calls to/from any other VoIP systems

NETWORK TOPOLOGIES

- Support for arbitrary network topologies
- Adjusts automatically to network split and merge

SUPPORTED CLIENTS

- **Bittium Tough VoIP™ products**
- COTS VoIP phones and soft-phones (for example Cisco, Linksys and Linphone)

SUPPORTED CALL MANAGERS
(connectivity to external VoIP networks)

- Cisco Unified Communications Manager (CUCM)
- FreeSWITCH

SIGNALING PROTOCOLS

- SIP (RFC 3261) and SDP (RFC 2327 / 4566)

VOICE SIGNALING

- RTP (RFC 3550), RTCP (RFC 36551) and Telephone events (RFC 2833)

AUDIO CODECS

- G.711, GSM, Speex (pass through codecs for example G.729AB and MELPe 2400)

MANAGEMENT INTERFACE

- WEB GUI management interface
- Configuration via XML configuration file
- One configuration file for all Tough VoIP Service nodes

SYNCHRONIZATION METHOD

- Between different multicast areas with unicast gateways
- Automatic multicast based discovery / synchronization

USER MOBILITY SUPPORT

- Supports phone mobility between Tough VoIP Service nodes
- Supports user mobility between different VoIP phones (Affiliation and de-affiliation)

CALL FEATURES

- Affiliation and de-affiliation (hot-desking)
- User defined and predefined group calls with audio mixing or PTT controlled
- Broadcast and multichannel call
- Priority point-to-point and group calls with call pre-emption (5 classes)
- Compressed dialing
- More features on the roadmap, for example call transfer and forward

SUPPORTED NETWORK SIZE AND USERS

- Up to 1000 Tough VoIP Service nodes in the network
- Up to 200 VoIP users in one Tough VoIP Service node
- Up to 10 000 VoIP users in the Tough VoIP Service network

SUPPORTED PLATFORMS

- Computer with Linux OS
- **Bittium TAC WIN Tactical Router™**

Performance on **Bittium TAC WIN Tactical Router™, without transcoding**

SIMULTANEOUSLY, PER NODE

- > 50 active call sessions
- > 200 registered subscribers

Recommended Minimum PC configuration

OS

- Installation packages for Debian and Ubuntu available
- Portability to other Linux versions

RAM (RESERVED BY TVS)

- 64 MB (reserved) / 512 MB (virtual) 64-bit build
- 48 MB (reserved) / 256 MB (virtual) 32-bit build

PROCESSOR

- Intel i3u (for tens of channels with transcoding)

STORAGE

- 32 MB (includes runtime log files)

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