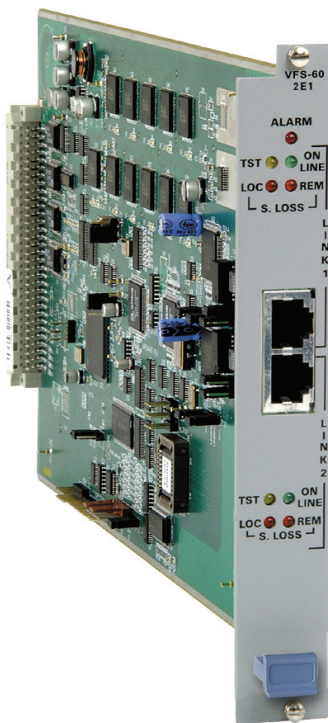


Megaplex-2100/2104

# VFS-24/30/48/60

Single/Dual E1/T1 Digital Voice Compression Server Modules



- Compression of 24/30/48/60 digital voice channels
- User-selectable compression rates: 8 kbps (G.729A), 5.3 kbps (G.723.1), or 6.4 kbps (G.723.1)
- On-board framer supporting ISDN PRI or T1/E1 PBX trunks, with DSU, CSU/LTU interface
- Voice activity detection, silence suppression, and comfort noise generation for efficient bandwidth utilization
- Analog voice compression

VFS modules connect and compress E1/T1 PBX voice trunks for efficient transmission over IP or TDM networks. Each single-slot VFS module provides one or two E1 or T1 ports (external ports), enhanced by one or two additional internal ports for server functionality. The timeslots received from the E1/T1 trunks or from the Megaplex backplane are compressed using standard algorithms such as G.723.1 (6.4 or 5.3 kbps per channel), or G.729A (8.0 kbps per channel). Compression methods are user-selectable per bundle.

The modules also act as voice and fax relay compression servers, compressing analog voice channels within the same module. A basic application is shown in *Figure 1*.

VFS modules are available in the following versions:

- **VFS-24** (single T1 trunk)
- **VFS-30** (single E1 trunk)
- **VFS-48** (two T1 trunks)
- **VFS-60** (two E1 trunks)

Since the VFS modules can compress and transmit an entire E1/T1 trunk using as little as 3 timeslots, Megaplex can utilize the remaining main link bandwidth to provide additional services such as data, Ethernet LAN, POTS and/or management, all on one platform (see *Figure 3*).

Multiple VFS modules enable Megaplex to compress and transmit as many as 300/240 digital voice channels (10 full E1/T1 trunks) over a single E1/T1 link. This results in very efficient utilization of E1/T1 or SDH/SONET networks.

To facilitate point-to-multipoint applications, voice timeslots can be grouped together into up to 8 separate bundles (internal ports). Up to 30 (E1 module) or 24 (T1 module) channels can be connected to each internal port. Each internal port can be routed to a different main link or IP destination (if using ML-IP). Likewise, each internal port can operate at a different voice encoding rate and/or fax rate.

Transfer up to  
300/240 compressed  
voice channels over a  
single E1/T1 link



data communications

The Access Company

# VFS-24/30/48/60

## Single/Dual E1/T1 Digital Voice Compression Server Modules

With regular TDM voice encoding methods alone, much bandwidth is “wasted” when normal periods of silence occur during a telephone call. VFS modules employ Voice Activity Detection (VAD) and silence suppression techniques, to maximize bandwidth utilization. After the G.723.1 or G.729A compression of the voice channels, silence suppression allows only channels transmitting actual conversation to fill main link timeslots; periods of silence are removed. At the remote side “comfort noise” is inserted to recreate the periods of silence that were removed, so that the quality of the call is not noticeably affected.

These methods use less bandwidth to transmit the same amount of voice without degrading the quality of the call. The user can select the amount of bandwidth (number of timeslots on the main link) that will be allocated per internal port, in accordance with the desired compression rate and the statistical periods of silence expected.

In addition to voice transmission, the VFS modules also perform automatic fax relaying, which allows the transmission of Group III fax (ITU-T Rec. V.17 and V.29) at rates of 2.4 to 14.4 kbps, irrespective of the digitizing rate selected for voice.

The modules have automatic rate fallback capability, to automatically switch to the next lower data rate supported by both communicating faxes. Voiceband modem transmissions at all the standard rates up to 14.4 kbps (per ITU-T Rec. V.22bis and V.32bis) are also supported, and are handled in the same way as fax transmissions.

Modem transmissions are handled as voiceband data.

Voiceband modem traffic at all the standard rates up to 14.4 kbps (per ITU-T Rec. V.22bis and V.32bis) are handled in the same way as fax transmissions.

The VFS modules can act as voice compression servers for voice modules, as shown in *Figure 2*.

The audio processing performed by the digital signal processors (DSPs) includes:

- Voice processing
- Processing of DTMF and call progress tones
- Processing of fax and voiceband modem signals.

Each DSP processes its group of timeslots independently of the other DSPs, in accordance with the parameters selected by the user for that group. However, within each group, the DSP detects the signal type (voice, DTMF, fax, etc.) carried in each timeslot (channel), and automatically selects the appropriate processing method.

A built-in adaptive G.168 echo canceller cancels the near-end hybrid echo. The echo canceller improves the voice quality on voice lines with long delay, such as long-distance calls or calls over non-terrestrial links (e.g., satellite). Echo delays of up to 16 msec are tolerated.

Diagnostics include loopbacks towards local and remote PBXs, and a local loop per timeslot bundle.

## Specifications

### Module Versions

VFS-24: single T1 port  
VFS-30: single E1 port  
VFS-48: two T1 ports  
VFS-60: two E1 ports

### Number of Internal Ports (Timeslot Bundles)

Up to 8 ports for regular operation  
Up to 2 additional ports for server operation

### Number of Voice Channels (Timeslots) per Internal Port

T1 modules: up to 24  
E1 modules: up to 30

### Voice Encoding Rate

Selectable per internal port:  
G.729A: 8 kbps per channel  
G.723.1: 6.4 kbps or 5.3 kbps per channel

### Fax Rates

Group III rates, selectable per internal port: 2.4, 4.8, 7.2, 9.6, 12.0, 14.4 kbps

### Bandwidth Allocation on Trunk

Selectable, according to the programmed voice encoding rate

### Acceptable Channel Bit Error Rate

$1 \times 10^{-3}$  or better

### Adaptive Echo Canceller

Tolerates delays of up to 16 msec per channel, as per G.168

### Silence Suppression

G.723.1A, G.729B

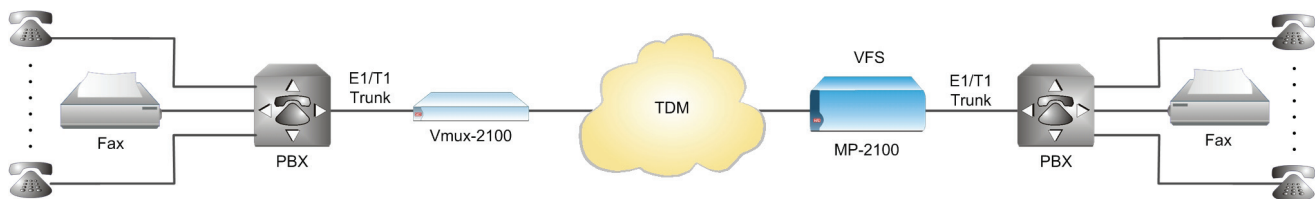


Figure 1. Transport of Compressed Digital Trunks between Megaplex and Vmux

**Signaling**

CAS, including R2 and E&M  
Transparent CCS

**Transparent Timeslot Mode**

Selected timeslots can be transmitted transparently for data or management relay

**Timing Modes**

INT Mode: Clock is provided by MP-2100 to the PBX

LBT Mode: Clock is provided by the PBX to MP-2100

**Diagnostics**

Auto self-test activated upon power-up and during normal operation

Local loopback towards the local PBX trunk

Remote loopback towards the remote PBX trunk

Local loopback on each internal port (bundle)

**Indicators**

ALARM (red) – On when fault detected on module

Per port:

ON LINE (green) – On when port is connected and functioning

TST (yellow) – On when test being performed on port

LOC/Red Alarm (red) – On when local sync loss is detected

REM (red)/Yellow Alarm (yellow) – On when remote sync loss is detected

**Configuration**

Via the Megaplex management system

**Power Consumption**

Module	Current [A]	Power [W]
VFS-24	1.0	5.0
VFS-30	1.2	5.8
VFS-48	1.6	8.0
VFS-60	1.7	8.3

**Note:** VFS modules utilize the +5 VDC line only.

**Environment**

Operating temperature: 0°C to 45°C  
(32°F to 113°F)

Storage temperature: -20°C to +70°C  
(-4°F to +160°F)

Humidity: up to 95%, non-condensing

**T1 INTERFACE MODULES****Voice Channels**

Up to 24 (per port)

**Data Rate**

1.544 Mbps (per port)

**Standards**

AT&T TR-62411, Pub. 54016, fANSI T1.403, ITU-T Rec. G.703, G.704

**Framing**

D4 (SF), ESF

**Line Code**

AMI

**Zero Suppression**

Software-selectable:

- Transparent (AMI coding - no zero suppression)
- B7ZS
- B8ZS

**Transmit Signal Level**

Nominal level:  $\pm 2.7V$  ( $\pm 10\%$ ), software-selectable, measured at 0-655 ft

**Receive Signal Levels**

0 to -10 dB without CSU

**Timing**

Internal or loopback

**Jitter Performance**

Per AT&T TR-62411

**Line Type**

Balanced 4-wire, 100 $\Omega$

**Connectors (per port)**

RJ-45

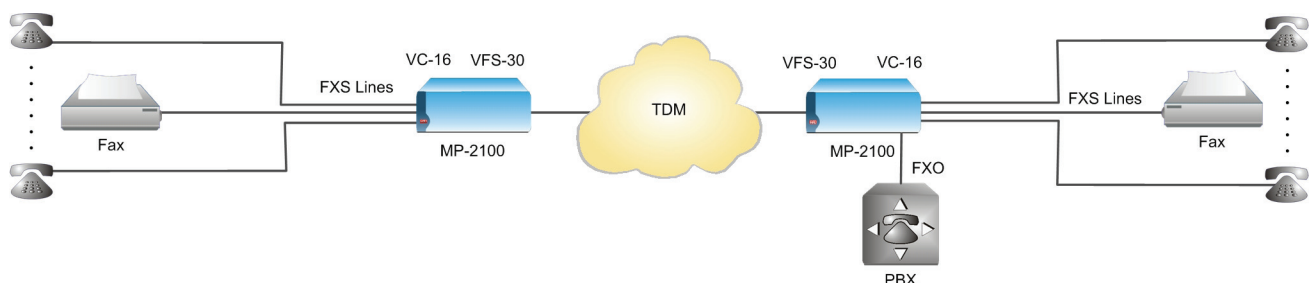


Figure 2. Transport of Compressed Digital and Analog Channels over TDM Network (VFS as Compression Server)

# VFS-24/30/48/60

## Single/Dual E1/T1 Digital Voice Compression Server Modules

### E1 INTERFACE MODULES

#### Voice Channels

Up to 30 (per port)

#### Data Rate

2.048 Mbps (per port)

#### Standards

ITU-T Rec. G.703, G.704, G.732

#### Framing

G.732N  
G.732N with CRC-4  
G.732S  
G.732S with CRC-4

#### Line Code

HDB3

#### Transmit Signal Level

Balanced:  $\pm 3V$  ( $\pm 10\%$ )  
Unbalanced:  $\pm 2.37V$  ( $\pm 10\%$ )

#### Receive Signal Levels

0 to -10 dB without LTU

#### Timing

Internal or loopback

#### Jitter Performance

Per ITU-T G.823

#### Line Type

Balanced: 4-wire, 120 $\Omega$   
Unbalanced: coax, 75 $\Omega$ ,  
(jumper-selectable)

#### Connectors (per port)

Balanced: 8-pin RJ-45

*Note: An MP-CBL-RJ45/2BNC/E1 adapter can be ordered to convert an E1 port RJ-45 connector into a pair of BNC connectors for unbalanced coax interface.*

### Ordering

#### MP-2100M-VFS-24/1T1

Single T1 Digital Voice Compression Server Module for MP-2100/2104

#### MP-2100M-VFS-30/1E1

Single E1 Digital Voice Compression Server Module for MP-2100/2104

#### MP-2100M-VFS-48/2T1

Dual T1 Digital Voice Compression Server Module for MP-2100/2104

#### MP-2100M-VFS-60/2E1

Dual E1 Digital Voice Compression Server Module for MP-2100/2104

### OPTIONAL ACCESSORIES

#### MP-CBL-RJ45/2BNC/E1

Adapter for VFS-30/60 modules converting an E1 port RJ-45 connector into a pair of BNC connectors for unbalanced coax interface

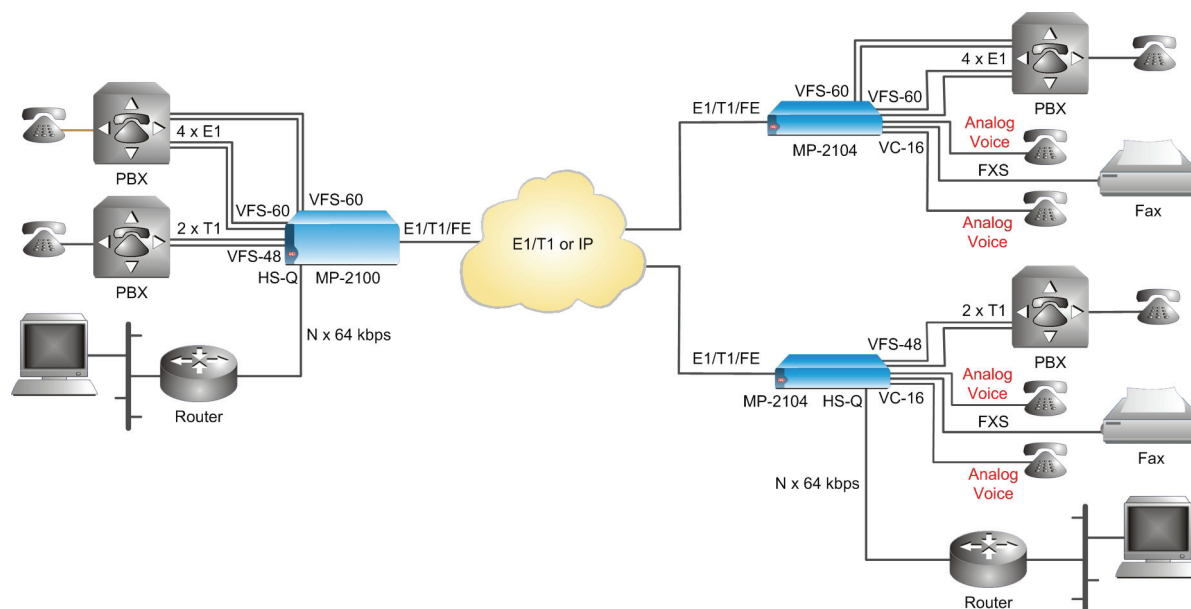


Figure 3. Combining Compressed Digital Voice, Analog Voice and Data in Point-to-Multipoint Topology

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