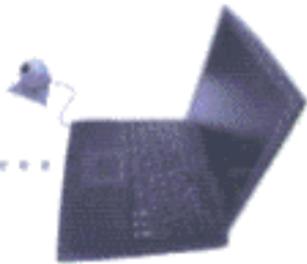


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IP TELEPHONY

DEPLOYING VOICE-
OVER-IP PROTOCOLS

 WILEY

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