MESSAGEmanager Solutions

MESSAGEmanager and IP Telephony

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Definitions & Terminology

Introduction to VoIP

Voice over IP refers to the ability to transmit voice and fax telephone calls over IP networks such as the internet using packet linked routes with a suitable quality of service.

There are currently two major international groups defining standards for VoIP:-

- International Telecommunications Union (ITU) which has defined
    - H.323 an umbrella specification for the implementation of packet based media over IP networks that cannot guarantee Quality of Service.
    - H.248 standard often referred to as the Megaco Protocol
- Internet Engineering Task Force (IETF) which has defined drafts of:-
    - RFC 3261 Session Initiation Protocol (SIP)
    - RFC27053 Media Gateway Control Protocol (MGCP).

H.323 was developed in the mid 1990s and is more mature than any of the above protocols.

SIP is an emerging protocol for setting up telephony, conferencing, multimedia and other types of communication sessions on the internet.

System Solutions MESSAGEmanager utilises Dialogic Host Media Processing (HMP) Software to provide flexible, scalable, and cost-effective next-generation IP media servers.

HMP software is an Intel® communications building block technology. When installed on a system, the software performs like an Intel® NetStructure™ board with DM3 architecture to the customer application, but all media processing takes place on the host processor.

HMP uses a network interface card (NIC) to provide IP connectivity. HMP supports the industry standard H.323 and SIP protocols for call control, and RTP/RTCP for media streaming over IP in G.711, G.723.1, G.729A, and G.729AB format.

HMP software is implemented as a Windows* operating system kernel-mode driver that runs at real-time priority. The software is optimized to run on the Intel® Pentium® III, Pentium® 4, and Xeon® processors. Since HMP software is implemented as a software-only product, it can be installed and upgraded like any other software. The HMP software is licensed using an industry-standard model in which the MAC address (NIC) is used to node-lock the software to a specific computer. To enable customers to choose combinations of media processing, the HMP media software is available through a flexible licensing scheme.

Features include:
• Global Call support for H.323 and SIP call control
• H.450.2 Call Transfer (H.323)
• RFC 2833 support
• H.245 User Input Indication (UII)
• IP Multicast (Transmit)
• Ability to scale up to 120 channels of media processing
• Low bit rate (G.723.1, G.729A, and G.729AB) coders for IP transcoding
• Speech Integration (Continuous Speech Processing)
• Support for a variety of media processing functions such as play with volume control, record with automatic gain control, user-defined tone detection and generation

Configuring Media and Interface Resources

With HMP software v1.1, resource provisioning for telephony solutions is done according to a flexible software license model rather than in fixed quantities as delivered in hardware boards. Choosing a customized “package” of resources for each license based on the types of resources available and the rules for combining them is a simple task.

This document aims to explain HMP software license configuration as succinctly as possible.

Types of Resources

Two general classes of resources are available in Version 1.1 of HMP software: media and interface.

Media Resources

Media resources are used for applications such as voice mail or interactive voice response (IVR), and currently there are four types:

- Voice Control functions such as play/record, tones, call progress, etc.
- Continuous Speech Processing (CSP) as an add-on to voice resources and required for speech enablement. See *Rules for Combining Resources* below.
- Conference. (Not Supported in MESSAGEmanager R2).
- T.38 Fax. (Not Supported in MESSAGEmanager R2).

IP Network Interface Resources

Interface resources are organized in two categories:

- IP media streaming
- IP call control

Note: Currently, the only network interface supported by HMP software v1.1 is the Internet Protocol (IP), and all network interface resources are used to provide IP connectivity.

IP Media Streaming Resources

IP media streaming resources provide the functionality required to stream media between HMP software and IP clients using the Real Time Transport Protocol (RTP). Two types of licensable resources are offered with HMP software:

- **Basic RTP G.711** — Acts as a “pipe” to establish an RTP connection between HMP software and one remote IP endpoint. Each Voice over IP (VoIP) call into the server running HMP software will require one RTP resource. This resource provides RTP streaming using the ITU-T G.711 voice coder.
- **Enhanced RTP** — Enables customers to implement the low-bit-rate coders (G.723 and G.729ab) for streaming over RTP. A Basic RTP G.711 resource is required for each Enhanced RTP to become operational.

See *Rules for Combining Resources* below.

IP Call Control Resource

IP Call Control resource is required to access IP call control functionality provided through the H.323 and SIP protocol stacks, which are included with HMP software. If IP Call control resources are licensed they must be used on all IP connections. For example on a system with 60 licensed Basic RTP G.711 resources, 60 Call Control resources must be licensed.

Summary of Available Resources

In summary, there are seven types of resources available with HMP software:

Media

- Voice
- Speech
- Conference
- T. 38 Fax

IP Network Interface

- Basic RTP G.711
- Enhanced RTP
- IP Call Control

Rules for Combining Resources

Two types of resources cannot be used alone, but must be combined with other resources. These are the Speech resource and the Enhanced RTP resource. Here are the rules:

- Each Speech resource requires a Voice resource.
- Each Enhanced RTP resource requires a Basic RTP G.711 resource.

Because speech is a way of enhancing a simple play/record voice function, each Speech resource needs a Voice resource.

Likewise, HMP software presumes the use of the G.711 voice coder. Allowing the use of low-bit-rate coders is an enhancement, since MP software defaults to G.711 when no low-bit-rate coder is in use. For this reason, each Enhanced RTP resource requires a Basic RTP G.711 resource.

Combining Resources to Create Basic Sessions

HMP software resources are combined to enable various sessions in an application just as they would if telephony boards were in use. Two common examples of sessions are provided here: a voice mail session, a speech enabled IVR session.

Voice Mail Session Example

A single voice mail session using G.729 and the delivered protocol stack requires:

- A Voice resource to provide media resources for play/record
- A Basic RTP G.711 resource
- An Enhanced RTP resource for G.729 support
- An IP Call Control resource

Speech-Enabled IVR Session Example

A single speech-enabled IVR session using G.711 and the delivered protocol stack, requires the following resources:

- A Voice resource to provide media resources for play/record
- A Speech resource for speech enablement via CSP
- A Basic RTP G.711 resource
- An IP Call Control resource

Combining Resources to Create Sessions for a Call Center

For a call center, two kinds of sessions are basic: RTP sessions in which a customer calls an agent or an agent calls another agent and IVR or auto-attendant sessions.

RTP Session Using G.729 Example

A single voice call session using G.729 requires four resources:

- A Voice resource to provide a media resource for the call
- A Basic RTP G.711 resource
- An Enhanced RTP resource for G.729 support
- An IP Call Control resource

RTP Session Using G.711 Example

A single voice call session using G.711 requires three resources:

- A Voice resource to provide media resources for the call
- A Basic RTP G.711 resource
- An IP Call Control resource

IVR/Auto-Attendant Session Using G.711 Example

A single IVR/auto-attendant session using G.711 requires the following resources:

- A Voice resource to provide media resources for the call to the IVR/auto attendant
- A Basic RTP G.711 resource
- An IP Call Control resource

Since the IVR/auto attendant is not speechenabled, only a Voice resource is required.

The number of Enhanced RTP resources governs the number of RTP sessions that will allow the use of low-bit-rate coders and lower bandwidth usage. The number of Enhanced RTP resources must be equal to or less than the number of Basic RTP G.711 resources. The number of IP Call Control resources must equal the number of Basic RTP G.711 resources.

Provisioning Examples

Two common types of solutions in which HMP software is used are unified messaging servers and IP call centers. Some hypothetical examples for provisioning such solutions are given here.

Unified Messaging Server Example

Table 1 provides the requirements for a unified messaging server matched with the HMP software resources required.

Scenario Requirements	Resource Requirements
A unified messaging server allowing 120 sessions, 60 of which will allow the use of low-bit-rate coders	120 Basic RTP G.711 resources, and 60 Enhanced RTP resources
and	120 IP Call Control resources
A maximum of 30 users of voice applications such as IVR and voice mail, 10 of which are speech-enabled	30 Voice resources, and 10 Speech resources

Table 1. Unified Messaging Server Requirements

See Figure 2 for an illustration of how these resources are grouped.

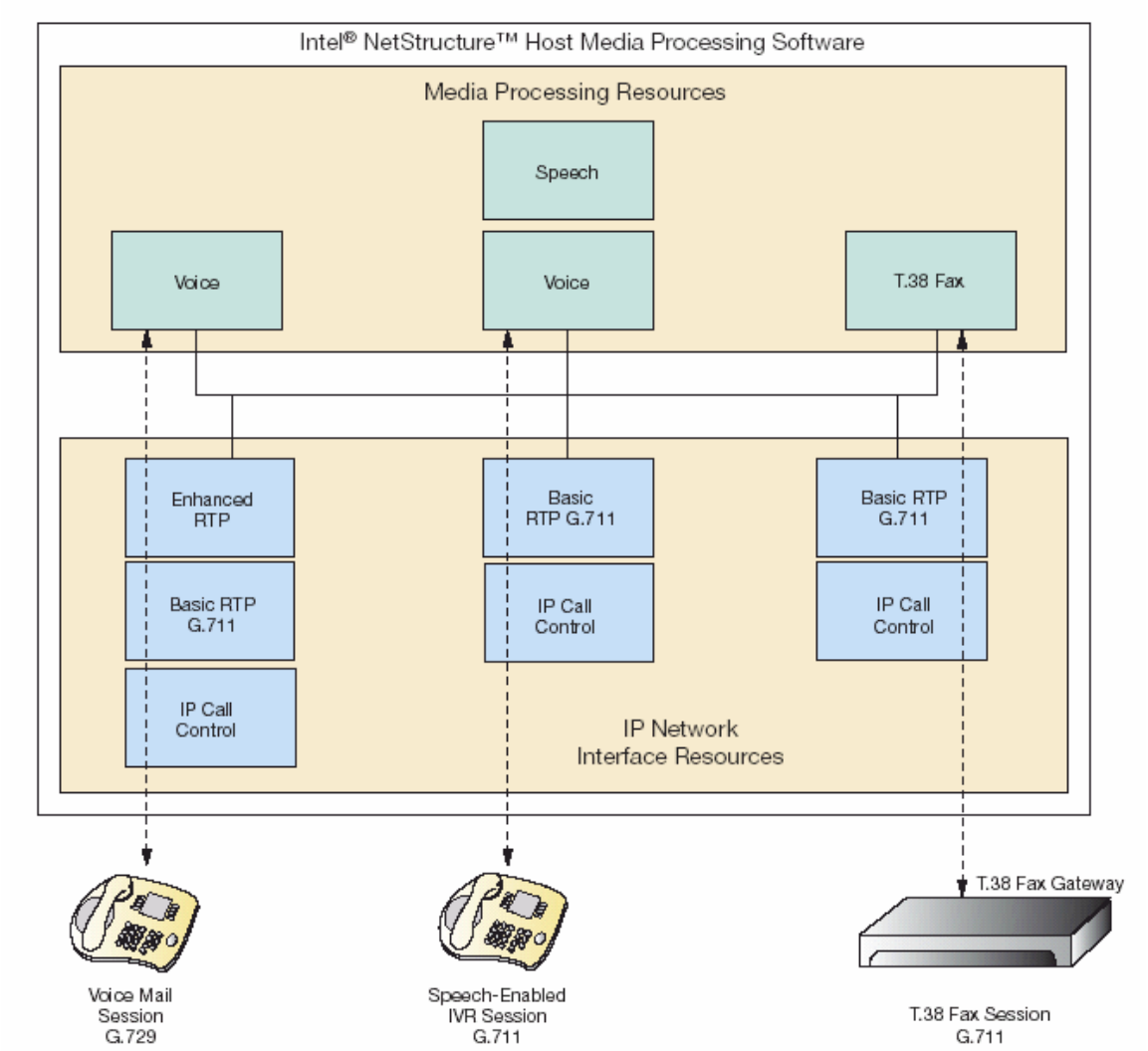

Figure 2. Resources for a Unified Messaging Server

IP Call Center Example

Table 2 provides the requirements for an IP call center matched with the HMP software resources required.

Scenario Requirements	Resource Requirements
An IP call center with 40 agents and 80 trunk lines for incoming calls, requiring 120 sessions, 60 of which will allow the use of low-bit-rate coders	120 Basic RTP G.711 resources with 60 Enhanced RTP resources
and	120 IP Call Control resources
A maximum of 60 voice ports with 30 possible speechenabled sessions	60 Voice resources, and 30 Speech resources

Table 2. IP Call Center Requirements

See Figure 3 for an illustration of how these resources are grouped.

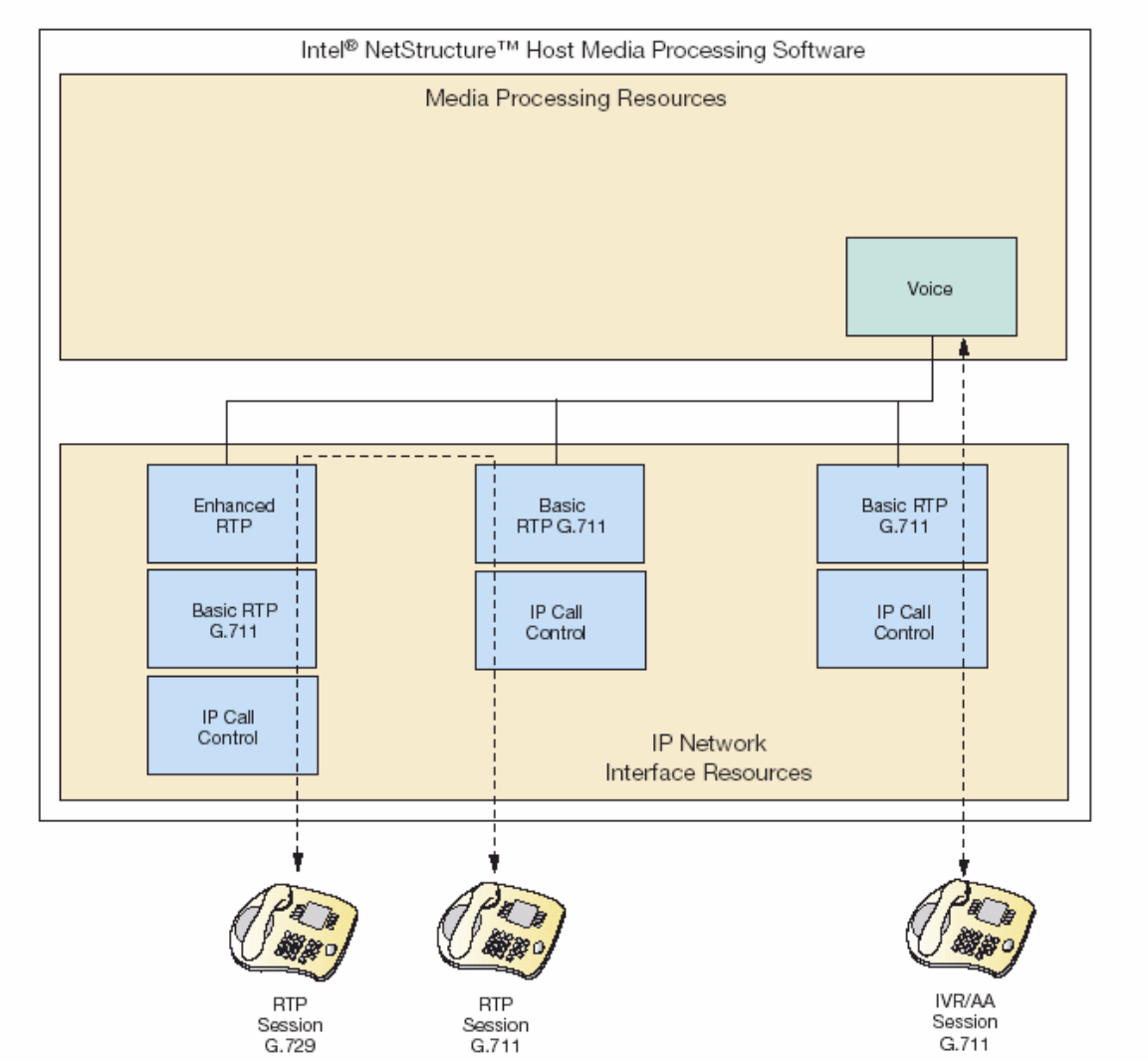

Figure 3. Resources for an IP Call Center

Checklist of Provisioning Questions

1. How many RTP sessions are needed?
2. How many Enhanced RTP sessions should be enabled for the use of a low-bit-rate coder?
3. How many Voice sessions will be allowed?
4. How many Voice sessions will be speechenabled?

Limitations

The following limitation is enforced per HMP software v1.1 license.

- No more than 64 Enhanced RTP resources per license

Summary Table

Table 3 provides a convenient summary of all the information required for configuring licenses for HMP software.

Resource Type	Limits	Notes
Basic RTP G.711	120 maximum; see “Total per License” below	Sum of the number of sessions allowed or lines plus trunks allowed; G.711 is the default coder used
Enhanced RTP	64 maximum; less than or equal to Basic RTP G.711	Allows use of low-bit-rate coder; must be equal to or less than the number of Basic RTP G.711 resources
IP Call Control	Equal to number of Basic RTP G.711	Requires one resource for each Basic RTP G.711 resource. Delivered IP call control stack (H.323 and SIP) must be used.
Voice	120 maximum; see “Total per License” below	Equals the number of play/record
Speech	sessions Less than or equal to Voice	Equals the number of Voice sessions that can be speech-enabled; must be equal to or less than the number of Voice resources

Table 3. Summary Table

PC Hardware System Requirements

The basic requirements to install and run this release are described in the following sections:
• Hardware Requirements
• Software Requirements
• IP Endpoints

Hardware Requirements

The minimum hardware requirements for this Release are:
• Intel® Pentium® III Processor (See Table 1. for processor recommendations.)
• CD-ROM drive
• VGA display
• Pointing device
• 100 BaseT Network Interface Card (NIC)

NOTE: *A 1000 BaseT NIC will improve CPU utilization.*

Number of User Sessions (IVR Only)	Minimum Processor Type and Clock Speed	
	G.711 (20 msec Frame)	G.723.1, G.729A, or G.729AB (on 50% User Sessions)
Up to 4	Intel® Pentium® III, 850 MHz	Intel® Pentium® III, 850 MHz
Up to 32	Intel® Pentium® III, 1.26 GHz	Intel® Pentium® 4, 1.7 GHz
Up to 64	Intel® Pentium® 4, 2.0 GHz	Dual Intel® Xeon™, 2.0 GHz
Up to 96	Single Intel® Xeon™, 2.4 GHz	Dual Intel® Xeon™, 2.8 GHz
Over 96	Dual Intel® Xeon™, 2.4 GHz	Dual Intel® Xeon™, 3.06 GHz

Table 1. Processor Recommendations

Hyper-Threading Technology

Hyper-Threading Technology (HT) is only supported on systems using the Intel® Pentium® 4 or Xeon™ processors and the Microsoft® Windows® 2000 or Windows® XP operating system.

For detailed information about using HT with the Windows operating system, see the following Intel® Web sites:
http://www.intel.com/technology/hyperthread/
http://www.intel.com/homepage/land/hyperthreading_more.htm

Also, refer to the following Microsoft Web site:
http://www.microsoft.com/windows2000/server/evaluation/performance/reports/hyperthread.asp
and the Microsoft design information titled “Windows Support for Hyper-Threading Technology.”

Memory Requirements

For production purposes, a minimum of 512 MB of memory is required.

Software Requirements

The software requirements are:
• Operating System:
- Windows 2000 with Service Pack 3 (Professional and Server)
or
- Windows XP Professional with Service Pack 1

Reference Configurations Tested

For HMP Release 1.1, the reference configurations listed in Table 3 have been successfully tested with CPU utilization of 50% or less:

Reference Configuration	Processor Type and Speed	Memory
4r-4e-4v-4c-4f-4s	Intel® Pentium® III Single CPU – 850 MHz Intel® Pentium® III Single CPU – 1.26 GHz	512 Mb
16r-8e-16v-16c-16f-16s	Intel® Pentium® III Single CPU – 1.26 GHz	512 Mb
32r-32e-32v-32c-32f-32s	Intel® Pentium® III Dual CPU – 1.26 GHz Intel Pentium P4 – 2.0 GHz	1 Gb
64r-64e-64v-64c-32f-64s	Intel® Xeon™ Dual CPU‡ – 2.6 GHz Intel® Xeon™ Dual CPU‡ – 2.8 GHz	512 Mb
96r-32e-96v-60c-10f-96s	Intel® Xeon™ Dual CPU – 2.8 GHz	512 Mb
96r-48v-96c	Intel® Xeon™ Single CPU – 2.8 GHz	1 Gb
96r-64e-96v	Intel® Xeon™ Dual CPU – 2.8 GHz	1 Gb
120r-120v-120s	Intel® Xeon™ Dual CPU – 2.6 GHz Intel® Xeon™ Dual CPU‡ – 2.8 GHz	1 Gb
120r-14v-120c	Intel® Xeon™ Single CPU – 2.8 GHz	1 Gb
‡ = Hyper-threading enabled on both processors = Dual CPU machine with one CPU disabled		

Table 3. Tested Reference Configurations

Refer to the Intel ® NetStructure ™ Host Media Processing Software Release 1.1 for Windows Installation Guide for instructions on obtaining and using a license for a specific HMP configuration.

Platforms Used in Testing the HMP 1.1 Software

Table 4 provides information about the chassis/platform configurations used to test the HMP 1.1 software.

Vendor	Processor	Processor Speed	Symmetric Multi-processing	Operating System
Transduction Ltd.	Pentium® III	850 MHz	No	Windows* 2000 Advanced Server, SP2
IBM Corp.	Pentium® III	1.26 GHz	No	Windows* 2000 Advanced Server, SP2
Transduction Ltd.	Pentium® 4	2.0 GHz	No	Windows* 2000 Advanced Server, SP2
Intel® TSRLT2 Carrier Grade Server	Pentium® III	1.26 GHz	No	Windows* 2000 Advanced Server, SP2
IBM Corp.	Pentium® 4/ x305 Series	2.0 GHz	No	Windows* 2000 Professional, SP3
IBM Corp.	Dual Xeon™/ x335 Series	2.4 GHz	Yes	Windows* XP Professional, SP1
Dell Inc.	Dual Xeon™	2.6 GHz	Yes	Windows* 2000 Advanced Server, SP3
Dell Inc.	Dual Xeon™	2.8 GHz	Yes	Windows* 2000 Advanced Server, SP3
Dell Inc.	Dual Xeon™	2.8 GHz	Yes	Windows* XP Professional, SP1

Table 4. Platform Configurations Used in Testing HMP 1.1 Software

Appendix A

G.711	ITU standard that describes the classic non-compressed 64 Kbps PCM voice coding technique for DS-Os running over the PSTN.
H.248	An ITU-T recommendation, formerly called Megaco by the IETF. H.248 is a control protocol used by Media Gateway Controllers (MGCs) to control Media Gateways (MGs). H.248 is essentially an extension of MGCP, to handle various forms of communication including voice, video and data.
H.323	Extension of ITU-T standard H3.20 that enables multimedia (real-time voice, data, video communication) over LANs and other packet-switched networks, as well as video conferencing over the Internet.
HT (Hyper-Threading Technology)	Circuitry added to a processor that enables it to appear as two logical processors, resulting in a single physical processor appearing like two logical processors to an operating system and multi-threaded application. Each logical processor can execute a thread of a multi-threaded program.
IETF (Internet Engineering Task Force)	The standards and specifications review board for the internet. The IETF defines standard Internet operating protocols.
IP	The 'IP' in Voice-over-IP. An open networking protocol by which data is sent from one computer to another on the Internet. In the OSI (Open Systems Interconnection) communication model, IP occupies Layer 3, the Networking Layer. Each user's endpoint node (a computer known
IP Multicast	A system for sending IP transmissions from one host out to many multiple users. This uses less bandwidth than would normally be required for audio and video broadcasting over the Internet.
ISO	Also know as the International Organisation for Standardisation and the International Standards Organisation. However, according to ISO, 'ISO' is not an abbreviation. It is a word, derived from the Greek *isos*, meaning 'equal', which is the root for the prefix 'iso-'. ISO defines international communications and information exchange standards.
ITU (International Telecommunications	The international standards body that helps and defines emerging standards. A charter organisation of the United Nations. The ITU is the parent organisation for ITU-T.
ITU-T (International Telecommunications Union, Telecommunications Standardization Sector)	One of the 3-Sectors of the ITU (International Telecommunications Union). The ITU-T's mission is to ensure an efficient and on-time production of high quality standards covering all fields of telecommunications.
IVR (Interactive Voice Response)	A computer telephony application that presents the an audio menu to the caller, then accepts a combination of voice telephone input and/or touch-tone keypad selection and provides responses in the form of voice prompts, fax, callback, email or other media. IVR is usually a component of a larger application that almost always includes database access. Common IVR applications include: bank, credit card and stock account balances and transfers, surveys and pools.

MAC (Media Access Control)	Lower sublayer of the OSI Reference Model layer 2, the Data-Link Layer. Controls the interface between the switch fabric and the PHY and thus controls access to a transmission medium. Also the portion of the Data-Link Layer that controls addressing of packets and enables data to be sent and received across a LAN.
MGCP (Media Gateway Control Protocol)	Also called the Media Gateway Controller Protocol. A protocol designed to bridge between the circuit-based PSTN and 'next-gen' IP based networks. A standardised protocol used for sending call-setup, tear-down, and related commands from the Call Agent to the MG (Media Gateway). It's competitor for a time was called H.248 by the ITU and Megaco by the IETF. MGs receive commands from the Call Agents, usually according to the MGCP or Megaco/H.248 standard. MGCP is somewhat popular in Asia, but much of the future of VoIP signaling belongs to SIP and Cisco's SCCP (Skinny Client Control Protocol).
NIC (Network Interface Card)	Also called an Adapter or Adapter Card. A circuit board (or chip) installed in a computer system, usually a PC, with an associated connector so that it can be connected to a LAN or WAN.
OSI (Open Systems Interconnect)	A reference model for communications systems created by the ISO and the ITU-T. It describes a 7-layer structure for modeling the interconnection and exchange of information between users in a data communications system. The 7 layers are: the Physical Layer, the Link layer (or Data Link Layer), the Network Layer, the Transport Layer, the Session Layer, the Presentation Layer and the Application layer. Its purpose is to guide product implementers so that their products will consistently work with other products.
RTP (Real-Time Transfer Protocol)	A protocol commonly used in IP networks to provide end-to-end transport to support delay-sensitive, real-time multimedia traffic over multicast or unicast network services.
SIP (Session Initiation Protocol)	A text-based Open Systems Interconnect (OSI) Application Layer protocol defined by the IETF for handling multimedia communications over IP networks. SIP is able to transport call setup, routing, authentication, termination and other feature messages to endpoints within the IP domain, whether those message originate from outside the IP cloud over PSTN resources or within the cloud. SIP has its origins in HTTP and is compatible with VoIP-related protocols, such as H.323, MGCP and H.248. SIP sports 5 major features: 1) User Location (determination of the end system to be used for communication). 2) User Capabilities (determination of the media and media parameters to be used. 3) User Availability (determination of the willingness of the called party to engage in communications). 4) Call Setup ("ringing", establishment of call parameters at both called and calling party. 5) Call Handling (including transfer and termination of calls).
VoIP (Voice over Internet Protocol)	A technology for transmitting ordinary telephone calls over IP networks (such as the Internet) using packet-linked routes.

Founded in 1982, MESSAGEmanager Solutions has developed a reputation for innovation in technology aligned with real world business strategies. MESSAGEmanager Solutions is the developer of MESSAGEmanager, a modular messaging and communications platform that integrates Voice Messaging, FAX, SMS, Telex, Speech Enabled IVR and Presence into Telephone Systems and Desktop and Back Office applications.

MESSAGEmanager Solutions is a Microsoft Gold Certified Partner, IBM Business Partner, SAP Certified Solution Partner, HP Development Partner, Intel, Cantata Technology, Cisco, Nuance, Alcatel, Avaya, DocsCorp and AudioCodes Partner.

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